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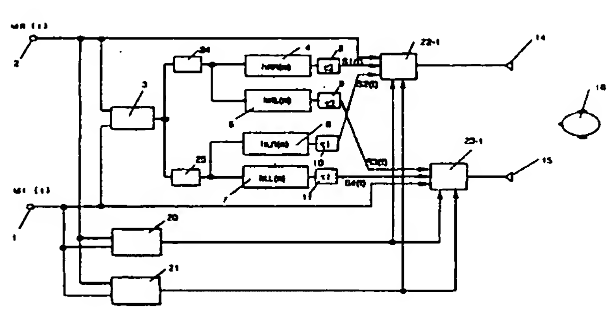
(54) Sound field controller

(57) A sound field controller for reproducing a sound field with presence comprises:

input means (1, 2) for inputting an input audio signal having two channel signals (MR(t), ML(t)),
signal extracting means (3) for receiving and processing the input audio signal, and producing an extracted signal from the input audio signal,
delay means (19-1, 19-2; 8, 9, 10, 11) for delaying the extracted signal by a predetermined time, and producing a delayed signal,
signal judging means (20) for receiving the input audio signal and judging whether the input audio signal is a voice signal or a non-voice audio signal to output a detecting signal indicating the result,
correlation determining means (21) for determining correlation ratio between the two channel signals of the input signal to output a determining signal,
adding means (22, 23; 22-1, 23-1) for receiving the input audio signal, the delayed signal, the detecting signal, and the determining signal, adding the input audio signal and the delayed signal with a predetermined summation ratio based on the detecting signal and the determining signal, and producing a resulting summed signal, and
output means (14, 15) for reproducing the summed signal.

Fig. 11



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Description

BACKGROUND OF THE INVENTION

1. Field of the Invention:

The present invention relates to a sound field controller for reproducing sound effects for use in audio equipment or in audio-visual (AV) equipment.

2. Description of the Related Art:

In recent years, as VTRs (video tape recorders) have become a common household item, a large-screened display and a sound reproduction system giving a sense of presence are desired to enjoy music recorded on recording media or programmed in softwares as well as movies on video tapes at home, thereby giving rise to the requirement of corresponding hardware development.

A conventional sound field controller will be explained with reference to the figures.

Figure **20** shows a hardware block diagram indicating the structure of a conventional sound field controller. Stereo-audio signals are input via input terminals **1** and **2** to the sound field controller. The conventional sound field controller comprises a multiplier **62** for multiplying an input signal by **-1**, an adder **63** adds the input signals, a delay circuit **64** for delaying the input signal by a predetermined time, adders **12-5** and **13-5** for adding the input signals, a multiplier **65** for multiplying the input signal by **-1**, and speakers **14** and **15** for reproducing the signals and playing the sound for a listener **16** facing the speakers **14** and **15.** ML(t) and MR(t) represent a Left-channel signal and a Right-channel signal of the stereo-audio signal respectively, and t represents a continuous time, ML(t) and MR(t) being functions of time. τ_3 represents the delay time in the delay circuit **64.**

The operation of the conventional sound field controller configured as above will be explained with reference to Figure **20.**

ML(t) is applied through the input terminal **1**, and MR(t) through the input terminal **2.** Each of the signals ML(t) and MR(t) thus input is divided into two parts, so that MR(t) is inputted to the adders **63** and **12-5** and ML(t) to the multiplier **62** and the adder **13-5.** The multiplier **62** multiplies ML(t) by **-1**, and the result -ML(t) is applied to the adder **63.** The adder **63** adds MR(t) and -ML(t) to produce the result MR(t)-ML(t), which is applied to the delay circuit **64.** The delay circuit **64** delays MR(t)-ML(t) by fixed time and produces MR(t-τ_3)-ML(t-τ_3). The output signal of the delay circuit **64** is divided into two branches of signal. One signal is applied to the adder **12-5,** and the other signal to the multiplier **65.** The multiplier **65** multiplies MR(t-τ_3)-ML(t-τ_3) by **-1**, and the result of multiplication, -(MR(t-τ_3)-ML(t-τ_3)), is applied to the adder **13-5.** The adder **12-5** adds MR(t) and MR(t-τ_3)-ML(t-τ_3), and the sum MR(t)+MR(t-τ_3) -ML(t-τ_3) is produced and output from the speaker **14**. The adder **13-5** adds ML(t) and -(MR(t-τ_3)-ML(t-τ_3)), and the result ML(t)-(MR(t-τ_3)-ML(t-τ_3)), is output from the other speaker **15.**

In this process, the signals MR(t-τ_3)-ML(t-τ_3) and -(MR(t-τ_3)-ML(t-τ_3)) in antiphases each other are mixed with the respective input signals and reproduced from the two speakers respectively, with the result that a sound field is generated with a non-identifiable localization or the sound image (or, the subtracted signals cancel the crosstalks thereby to yield the feeling as if the right and left signals are reproduced from outside of the two speakers). By adjusting the mix balance with ML(t) and MR(t) which are unprocessed direct sound signals, a sound field is produced with expansion and presence (i.e. the sound is produced with giving a sense of expansion of the sound and a sense of presence to a listner). For example, a sound reproduction giving a listener the illusion of being in the same room (such as a concert-hall) as the original source of sound rather than in the room with the sound reproducing system is a sound reproduction with presence.

In the above-mentioned structure, however, adjustment of the sound field is performed by the mix balance between the antiphased sounds and direct sounds. Thus, if the antiphased sounds are relatively small, that would reduce the effect, while if the antiphased sounds are made larger to emphasize the effect, that would strengthen the antiphased sound, bringing about an uncomfortable feeling to the listener. Further, in the case where the input signal is a voice-sound signal, the conventional structure has a problem that the voice component is reduced when the difference signal of the input signal is added to the input signal, thereby the reproduced voice sound being ambiguous.

Figure **21** shows a block diagram of a conventional sound field controller capable of sound reproduction with presence.

In Figure **21**, input terminals **1** and **2** are supplied with a signal ML(t) to be reproduced from the left side channel (Lch) as viewed from the listener **16** and a signal MR(t) to be reproduced from the right side channel (Rch) as viewed from the listener **16,** respectively. These input terminals **1** and **2** are connected to speakers **74** and **75.** These two signals are added to each other by an adder **72** at a predetermined ratio, and then applied to a speaker **76** arranged at the front center of the listener **16.**

Also, the two signals ML(t) and MR(t) are processed and applied to a surround signal generation circuit **71**. The surround signal generation circuit **71** generates a signal S(t) called a surround signal indicting a reverberation and/or a reflection, which is caused when the input signal is output from the speakers in an ordinary room. The surround signal S(t) produced by the surround signal generation circuit **71** is applied to two speakers **69** and **70** arranged on the left and right sides of the listener **16**. The signals ML(t) and MR(t) normally represent what is called the stereo signal, or main signals as compared with the surround signal S(t).

In the structure shown in Figure **21**, the 2-channel (2ch) signals ML(t) and MR(t) normally reproduced from the VTR, etc. are applied to the surround signal generation circuit **71**. The surround signal generation circuit **71** generates the surround signal S(t) of the reverberation or the reflection. The main signals ML(t) and MR(t) are reproduced from the speakers **74** and **75** respectively, and the surround signal S(t) is divided into two parts and reproduced from the speakers **69** and **70**. Also, the main signals ML(t) and MR(t) are added at a predetermined ratio by the adder **72**, and the resulting sum signal is reproduced from the speaker **76**.

As compared with a 2ch stereo reproduction system generally using two front speakers, the above-mentioned audio reproduction system allows a sound reproduction with good presence by reproducing sounds that had been audible from the front only or sounds that could not be heard, from the sides or behind as a surround sound. Further, since the main signals ML(t) and MR(t) are added at an appropriate level and reproduced from the center speaker **76**, the front sound image is definitely localized.

In the above-mentioned structure, however, additional speakers arranged on the side or behind for reproducing surround signal are required as well as the space for accommodating the speakers.

In view of the problems of the conventional sound field controllers described above, the object of the present invention is to provide a sound field controller having a simple structure which is capable of unambiguous reproduction of a sound signal with presence and natural expansion.

Another object of the present invention is to provide a sound field controller for reproducing the sounds including the reflected and/or reverberation which are audible as if they are from positions other than the reproduction point of the speakers, thereby making possible a sound reproduction with presence without using any additional speakers on the sides or behind the listener.

SUMMARY OF THE INVENTION

A first sound field controller for reproducing a sound field with presence of this invention, comprises an input unit for inputting an input audio signal having a first and a second channel signals, a signal extracting circuit for receiving and processing the input audio signal, and producing an extracted signal of the input audio signals, an operation circuit for receiving the extracted signal from the signal extracting circuit, performing a convolution on the extracted signal, and generating a convolution sum signal, a delay circuit for delaying the convolution sum signal by a predetermined time, and producing a delayed signal, an adding circuit for receiving the input audio signal and the delayed signal, and adding the input audio signal and the delayed signal with a predetermined summation ratio to produce a summed signal, and an output circuit for reproducing the summed signal to localize a sound image in a desirable direction.

A second sound field controller for reproducing a sound field with presence according to the present invention, comprises; an input unit for inputting an input audio signal having two channel signals, a signal extracting circuit for receiving and processing the input audio signals, and producing an extracted signal of the input audio signals a delay circuit for delaying the extracted signal by a predetermined time, and producing a delayed signal, a signal judging circuit for receiving the input audio signals and judging whether the input audio signals are voice signals or a non-voice audio signal and to output a detecting signal indicating the result, a correlation determining circuit for determining correlation ratio between the two channel signals of the input signal to output a determining signal, an adding circuit for receiving the input audio signals, the delayed signal, the detecting signal, and the determining signal, adding the input audio signals and the delayed signal with a predetermined summation ratio based on the detecting signal and the determining signal, and producing a resulting summed signal, and an output unit for reproducing the summed signal.

A third sound field controller for reproducing a sound field with presence according the present invention comprises an input unit for inputting an input audio signal having two channel signals, a signal extracting circuit for receiving and processing the input audio signals, and producing an extracted signal of the input audio signals, a signal processing circuit for receiving the extracted signal, and for adding a reflected sound signal and/or a reverberaed signal signal to the extracting signal to produce a processed signal, an adding circuit for receiving the input audio signal and the processed signal, and adding the input audio signal and the processed signal with a predetermined summation ratio to produce a summed signal, and an output unit for reproducing the summed signal.

A fourth sound field controller for reproducing a sound field with presence according to the invention comprising an input unit for inputting an input audio signal having two channel signals, a signal processing circuit for receiving the input audio signals, and for adding a reflected sound signal and/or a reverberated sound signal to the input audio signal to produce a processed signal, an operation circuit for receiving the processed signal from the signal processing circuit,

performing a convolution on the processed signal, and generating a convolution sum signal, an adding circuit for receiving the processed signal and the convolution sum signal, and adding the processed signal and the convolution sum signal with a predetermined summation ratio to produce a summed signal, and an output unit for reproducing the summed signal to localize a sound image in a desirable direction.

In one embodiment of the present invention, the operation circuit comprises a first, a second, a third, and a forth operation portions, the delay circuit comprises a first, a second, a third, and a forth delay elements, each delay element receiving the convolution sum signal from the corresponding operation portion, and the adding circuit comprises a first and a second adders, the first adder receiving the first channel signal of the input signal and the delayed signal from the first and the third delay elements, the second adders receiving the second channel signal of the input audio signal and the delayed signal from the second and the forth delay elements.

In another embodiment of the present invention, the sound field controller further comprises a signal judging circuit for receiving the input audio signal and judging whether the input audio signal is a voice signal or a non-voice audio signal and to output a detecting signal indicating the result, a correlation determining circuit for determining correlation ratio between the two channel signals of the input signal to output a determining signal, wherein, the adding circuit further receives the detecting signal and the determining signal, and adjusts the summation ratio based on the detecting signal and the determining signal.

In another embodiment of the present invention, the sound field controller further comprises a signal processing circuit for receiving the input audio signal, adding a reflected sound signal and/or a reverberated sound signal to the input audio signal to produce a processed signal, and applying the processed signal to the operation circuit.

In another embodiment of the present invention, the operation circuit comprises a first and a second operation portions, the delay circuit comprises a first, a second, a third, and a forth delay elements, the first and the second delay elements receiving the convolution sum signal from the first operation portion, the third and the forth delay elements receiving the convolution sum signal from the second operation portion, and the adding circuit comprises a first and a second adders, the first adder receiving the first channel signal of the input signal and the delayed signal from the first and the third delay elements, the second adder receiving the second channel signal of the input audio signal and the delayed signal from the second and the forth delay elements.

In another embodiment of the present invention, the sound field controller further comprises a signal processing circuit for receiving the input audio signal, adding a reflected sound signal and/or a reverberated sound signal to the input audio signal to produce a processed signal, and applying the processed signal to the operation circuit, the signal processing circuit including a first processing part for the first and the second operation portions and a second processing part for the third and the forth operation portions.

In another embodiment of the present invention, the sound field controller further comprises a signal judging circuit for receiving the input audio signal and judging whether the input audio signal is a voice signal or a non-voice audio signal and to output a detecting signal indicating the result, a correlation determining circuit for determining correlation ratio between the two channel signals of the input signal to output a determining signal wherein, the adding circuit further receives the detecting signal and the determining signal, and adjusts the summation ratio based on the detecting signal and the determining signal.

A fifth sound field controller for reproducing a sound field with presence according to the present invention comprising an input unit for inputting an input audio signal having a first and a second channel signals, a signal extracting circuit for receiving and processing the input audio signal, and producing a sum signal and a difference signal of the first and second channel signals, a signal processing circuit for receiving the sum signal and the difference signal, and for adding a reflected sound signal and/or a reverbration signal to the sum signal and the difference signal to produce a processed signal, an adding circuit for receiving the input audio signal the processed signal, and adding the input audio signal and the processed signal with a predetermined summation ratio to produce a summed signal, an output unit for reproducing the summed signal.

In one embodiment of the present invention, the sound field controller further comprises a signal judging circuit for receiving the input audio signal and judging whether the input audio signal is a voice signal or a non-voice audio signal and to output a detecting signal indicating the judged result, a correlation determining circuit for determining correlation ratio between the two channel signals of the input signal to output a determining signal, wherein, the signal processing circuit includes a first processing portion for receiving the sum signal, and for adding a reflected sound signal and/or a reverbrated sound signal to the sum signal to produce a first and a second processed signals; and a second processing portion for receiving the difference signal, and for adding a reflected sound signal and/or a reverberated sound signal to the difference signal to produce a third and a forth processed signals, the adding circuit includes a first adder for deceiving the second channel signal and the first and the third processed signals, and for adding the second channel signal and the first and the third processed signals with a predetermined summation ratio to produce a first summed signal; and a second adder for receiving the first channel signal and the second and the forth processed signal, and for adding the first channel signal and the second and the forth processed signals with a predetermined summation ratio to produce a second summed signal, and the output circuit includes a first output portion for the first summed signal and

a second output portion for the second summed signal.

In another embodiment of the present invention, the sound field controller further comprises signal mixing circuit, wherein, the signal processing circuit includes a first processing portion for receiving the sum signal, and for adding a reflected sound signal and/or a reverberated sound signal to the sum signal to produce a first and a second processed signal; and a second processing portion for receiving the difference signal, and for adding a reflected sound signal and/or a reverberated sound signal to the difference signal to produce a third and a forth processed signals, the adding circuit includes a first adder for receiving the first and the third processed signals, and for adding the first and the third processed signals with a predetermined summation ratio to produce a first output signal; and a second adder for receiving the second and the forth processed signals, and for adding the second and the forth processed signals with a predetermined summation ratio to produce a second output signal, the signal mixing circuit receives the first and the second output signals, subtracts the second output signal from the first output signal with a predetermined subtracting ratio to produce a first summed signal, and adds the first output signal to the second output signal with a predetermined summation ratio to produce a second summed signal, and the output circuit includes a first output portion for the first summed signal and a second output portion for the second summed signal.

In another embodiment of the present invention, the A sound field controller further comprises a signal judging circuit for receiving the input audio signal and judging whether the input audio signal is a voice signals or a non-voice audio signal and to output a detecting signal indicating the result, a correlation determining circuit for determining correlation ratio between the two channel signals of the input signal to output a determining signal, wherein, the signal mixing circuit further receives the detecting signal and the determining signal, and adjusts the summation ratio and the subtracting ratio based on the detecting signal and the determining signal.

Thus, the invention described herein makes possible the advantages of (1) providing a sound field controller which reproduces a sound image including the reflection at a desirable position and direction without using any additional speakers on the sides or behind of the listener, and (2) providing a sound field controller in which the summation ratio of the surround signal (such as the reverberation and the reflection) and the input audio signal is appropriately adjusted so as to reproduce the surround signal effectively without making the main signal unclear.

These and other advantages of the present invention will become apparent to those skilled in the art upon reading and understanding the following detailed description with reference to the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure **1** is a hardware block diagram showing a sound field controller according to a first embodiment of the invention.

Figure **2** is a block diagram for explaining the principle of an operation circuit of a sound field controller according to the first embodiment of the invention.

Figure **3** is a diagram for explaining the structure of an operation circuit of a sound field controller according to the first embodiment of the invention.

Figure **4** is a hardware block diagram showing a sound field controller according to a second embodiment of the invention.

Figure **5** is a hardware block diagram showing a sound field controller according to a third embodiment of the invention.

Figure **6** is a diagram for explaining the principle of a signal decision circuit for a sound field controller according to the third embodiment of the invention.

Figure **7** is a hardware block diagram showing a sound field controller according to a fourth embodiment of the invention.

Figure **8** is a hardware block diagram showing a sound field controller according to a fifth embodiment of the invention.

Figures **9A** and **9B** are a diagrams for explaining the method of reflection addition for a sound field controller according to the fifth embodiment of the invention.

Figure **10A** is a block diagram for explaining the structure of a reflected sound generation circuit for a sound field controller according to the fifth embodiment of the invention.

Figure **10B** is a diagram showing a reflection series generated by the reflected sound generation circuit shown in Figure **10A**.

Figure **11** is a hardware block diagram showing a sound field controller according to a sixth embodiment of the invention.

Figure **12** is a hardware block diagram showing a sound field controller according to a seventh embodiment of the invention.

Figure **13** is a hardware block diagram showing a sound field controller according to an eighth embodiment of the invention.

Figure **14** is a hardware block diagram showing a sound field controller according to a ninth embodiment of the invention.

Figure **15** is a hardware block diagram showing a sound field controller according to a tenth embodiment of the invention.

Figure **16** is a hardware block diagram showing a sound field controller according to an 11th embodiment of the invention.

Figure **17** is a hardware block diagram showing a sound field controller according to a 12th embodiment of the invention.

Figure **18** is a hardware block diagram showing a sound field controller according to a 13th embodiment of the invention.

Figures **19A** is a diagram showing a reflection series generated by one reflected sound generation shown in Figure **18**.

Figure **19B** is a diagram showing a reflection series generated by another reflected sound generation circuit shown in Figure **18**.

Figure **19C** is a diagram for explaining the method of reflection addition for a sound field controller according to the 13th embodiment of the invention.

Figure **20** is a hardware block diagram showing a conventional sound field controller.

Figure **21** is a hardware block diagram showing another conventional sound field controller.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

The preferred embodiments of the present invention are described hereinbelow with reference to the accompanying figures.

Example 1

Figure **1** shows a block diagram of a sound field controller according to the first example of the present invention. The circuits having the same functions as the corresponding parts of the conventional field controller are represented by the same reference numerals as those in Figures **20** and **21** and will not be described in detail.

In Figure 1, a left-channel (hereinafter referred to as "Lch") signal ML(t) is applied to an input terminal **1** and a right-channel (hereinafter referred to as "Rch") signal MR(t) is applied to an input terminal **2.** These signals are divided into two branches respectively. One of the branched signals of ML(t) and one of the branched signals of MR(t) are applied to a difference signal extractor **3** and the others to adders **13** and **12** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto, and outputs the difference signal to operational circuits **4, 5, 6,** and **7.**

Each of the operational circuits **4** and **5** comprises an FIR filter having an impulse response, whereby the sound image being localized on the right side or right rear of the listener **16** by FIR filtering. Each of the operational circuits **6** and **7** comprises an FIR filter having an impulse response which allows the sound image to be localized on the left side or left rear of the listener **16** by convolution. In other words, the operational circuit **4** has an impulse response hRR(n), the operational circuit **5** an impulse response hRL(n), the operational circuit **6** an impulse response hLR(n), and the operational circuit **7** an impulse response hLL(n).

The output of the operational circuitry **4** is applied to the adder **12** via a delay circuit **8,** the output of the operational circuit **5** to the adder **13** via a delay circuit **9,** the output of the operational circuitry **6** to the adder **12** via a delay circuit **10,** and the output of the operational circuitry **7** to the adder **13** through a delay circuit **11.** The delay circuits **8** and **9** delay the input signals by the delay time τ_2, and the delay circuits **10** and **11** delay the input signals by the delay time τ_1. The adder **12** adds the signals output from the input terminal **2**, the delay circuit **8,** and the delay circuit **10** to each other at an arbitrary ratio. The adder **13** adds the signals output from the input terminal **1**, the delay circuit **9**, and the delay circuit **11** at an arbitrary ratio. The output signals of the adders **12** and **13** are applied to speakers **14** and **15** respectively. These signals are applied to the speakers **14** and **15** through respective power amplifiers (not shown in the figure) for amplifying the signals.

The operation of the sound field controller according to the first embodiment with above-mentioned structure will be explained below.

First, acoustic signals ML(t) and MR(t) of a voice, sound, or music is applied via the respective input terminals **1** and **2.** Each of the input signals are divided into two branches respectively. One of the branched signals of ML(t) and one off the branched signals of MR(t) are applied to a difference signal extractor **3** and the others to adders **13** and **12** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto, and outputs the difference signal to operational circuits **4, 5, 6,** and **7.**

In the difference signal calculated by the difference signal extractor **3,** the centrally-localized signal may be substan-

tially cancelled and most of the components would be reverberation components of Lch and Rch signals which are inserted during recording or broadcasting. For example, when the input signals are music signals with the singing voice of a singer, the centrally-localized signal of the singer's voice signal is almost canceled by subtracting operation with the remainder of reverberation components in the difference signal. For this reason, the difference signal is sometimes called a surround signal. The operational circuits **6** and **7** perform the convolution on the input signal to localize the sound image on the left side or left rear.

A method for virtually localizing the sound image in an arbitrary direction will be explained with reference to Figure **2**. Figure **2** shows a diagram indicating the principle of virtually generating a sound image localization using the Lch speaker **15** and the Rch speaker **14**, which is equivalent to a sound image localization generated from the signal reproduced from a left-side speaker **45**. In Figure **2**, the speakers **14** and **15** are located on the left and right sides respectively in front of the listener **16**. The input signal S(t) is applied to the operational circuits **6** and **7**. The operational circuit **6** comprises an FIR filter for performing convolution with impulse responses hLR(n), and the operational circuit **7** comprises an FIR filter for performing convolution with impulse response hLL(n). In the diagram, h1(t) represents the impulse response at the left-ear position (more accurately, the position of the eardrum, or in the case of measurement, the entrance of the acoustic meatus) of the listener **16** when the speaker **15** produces an impulse sound. Similarly, h2(t) represents the impulse response at the right-ear position of the listener **16** when the speaker **15** produces the impulse sound. Also, h3(t) represents the impulse response at the left-ear position when the speaker **14** produces an impulse sound, h4(t) represents the impulse response at the right-ear position of the listener **16** when the speaker **14** produces the impulse sound, h5(t) represents the impulse response at the left-ear position of the listener **16** when the speaker **45** produces the impulse sound, and h6(t) represents the impulse response at the right-ear position of the listener **16** when the speaker **45** produces the impulse sound.

In this configuration, when the signal S(t) is produced from the speaker **45**, the sound that reaches the ears of the listener **16** is expressed by the following equations:

Specifically, the sound pressure L(t) at the left ear is represented by Equation (1).

$$L(t) = S(t)*h5(t) \tag{1}$$

The sound pressure R(t) at the right ear is expressed as

$$R(t) = S(t)*h6(t) \tag{2}$$

where * represents a convolution.

A transfer function of the speaker itself which is practically to be multiplied is ignored in the case under consideration. Alternatively, the transfer function of the speakers may be considered to be included in the impulse response functions.

Further, supposing that the sound pressures L(t) and R(t) given by Equations (1) and (2), the impulse responses h1(t) to h6(t), and the signal S(t) are all temporally discrete digital signals, they are converted to the formations as shown by the following expressions (3), (4), (5), (6) and (7).

$$L(t) \rightarrow L(n) \tag{3}$$

$$R(t) \rightarrow R(n) \tag{4}$$

$$h5(t) \rightarrow h5(n) \tag{5}$$

$$h6(t) \rightarrow h6(n) \tag{6}$$

$$S(t) \rightarrow S(n) \tag{7}$$

In this case, Equations (1) and (2) are expressed by following Equations (8) and (9) respectively.

$$L(n) = S(n)*h5(n) = \sum_{k=0}^{N-1} S(k) \cdot h5(n-k) \tag{8}$$

$$R(n) = S(n)*h6(n) = \sum_{k=0}^{N-1} S(k) \cdot h6(n-k) \tag{9}$$

It would be noted that the natural number n should actually be expressed by nT instead, T indicating a sampling time. However, T is omitted as usual and Equations (8) and (9) are written in the above-mentioned expression.

Similarly, when the signal S(t) is reproduced from the speakers **14** and **15**, the sound which reaches the ears of the listener **16** is represented by following Equations (10) and (11). The sound pressure at the left ear is given by Equation (10).

$$L'(n) = S(n)*hLL(n)*h1(n) + S(n)*hLR(n)*h3(n) \tag{10}$$

The sound pressure at the right ear is expressed by Equation (11).

$$R'(n) = S(n)*hLL(n)*h2(n) + S(n)*hLR(n)*h4(n) \tag{11}$$

Assuming that the sounds are perceived as coming from the same direction if the head related transfer functions of the sounds are equivalent to each other (i.e. the direction from which sound is coming is determined based on the amplitude difference and the time difference between the sounds reaching the right and left ears, and this assumption is generally valid), Equations (12) to (15) hold as follows.

$$L(n) = L'(n) \tag{12}$$

$$h5(n) = hLL(n)*h1(n) + hLR(n)*h3(n) \tag{13}$$

$$R(n) = R'(n) \tag{14}$$

$$h6(n) = hLL(n)*h2(n) + hLR(n)*h4(n) \tag{15}$$

Thus, the impulse responses hLL(n) and hLR(n) may be determined so as to satisfy Equations (13) and (15).

The impulse responses h1(t) to h6(t) and hLL(t) to hLR(t) are rewritten in a frequency domain expression as shown by following Equations (16) to (23).

$$H1(n) = FFT(h1(n)) \tag{16}$$

$$H2(n) = FFT(h2(n)) \tag{17}$$

$$H3(n) = FFT(h3(n)) \tag{18}$$

$$H4(n) = FFT(h4(n)) \tag{19}$$

$$H5(n) = FFT(h5(n)) \tag{20}$$

$$H6(n) = FFT(h6(n)) \tag{21}$$

$$HLL(n) = FFT(hLL(n)) \tag{22}$$

$$HLR(n) = FFT(hLR(n)) \tag{23}$$

where FFT( ) represents a function transformed by Fourier transformation (FFT: Fast Fourier Transformer).

Next, Equations (13) and (15) are also rewritten in the frequency domain expression. The operation is transformed from a convolution to a multiplication as represented in Equations (24) and (25). The remaining parts are transformed to the transfer functions with the respective impulse responses by Fourier transformation.

$$H5(n) = HLL(n) \cdot H1(n) + HLR(n) \cdot H3(n) \tag{24}$$

$$H6(n) = HLL(n) \cdot H2(n) + HLR(n) \cdot H4(n) \tag{25}$$

In Equations (24) and (25), the values other than the transfer functions HLL(n) and HLR(n) are obtained by measurement. Therefore, the transfer functions HLL(n) and HLR(n) can be obtained from following Equations (26) and (27).

$$HLL(n) = \frac{H5(n) \cdot H4(n) - H6(n) \cdot H3(n)}{H1(n) \cdot H4(n) - H2(n) \cdot H3(n)} \tag{26}$$

$$HLR(n) = \frac{H6(n) \cdot H1(n) - H5(n) \cdot H2(n)}{H1(n) \cdot H4(n) - H2(n) \cdot H3(n)} \tag{27}$$

By using hLL(n) and hLR(n) obtained from HLL(n) and HLR(n) by preforming the inverse Fourier transformation (IFFT), ad applying the signal S(n) to the operational circuits **6** and **7**, the signal to be reproduced from the speaker **15** is obtained by performing the convolution with S(n) and hLL(n), and the signal to be produced from the speaker **14**, is obtained by preforming the convolution with S(n) and hLR(n). When the convolution sum signals are reproduced and the corresponding sounds are output from the respective speakers **14** and **15**, the listener can perceive the sounds as if the sound comes from the left speaker **45** that is not actually played.

The method described above can virtually localize the sound image in a desirable direction.

An exemplary structure of an FIR filter for performing convolution is shown in Figure **3**. In Figure **3**, the signal is applied to a signal input terminal **46** and goes through serially connected N-1 delay elements **47**. Each of delay elements **47** delays the signal by τ, each of multipliers **48** multiplies the input signal by a value called the tap (a coefficient of FIR filter) indicated by h(n), an adder **49** adds all the signals output from the multipliers **48**, and the added (sum) signal is output via an output terminal **50**. Although the FIR filter shown in Figure **3** is formed by hardware, the FIR filter may be implemented by using a DSP (Digital Signal Processor) or a custom LSI for high speed multiplication and addition operations.

The impulse responses h(n) (n: 0 to N-1, where N is the required length of the impulse response) are set up as the tap coefficients of the respective multipliers **48** as shown in Figure **3**. Also, a delay time corresponding to the sampling frequency of converting an analog signal to a digital signal is set up in each of the delay elements **47**. The signals applied to the input terminal **46** are multiplied/added/delayed repeatedly, thereby the convolution as shown in Equations (8) and (9) is performed. This operation involves digital signals. In practice, therefore, an A/D converter and a D/A converter are to be provided in order to convert analog signals to digital signals before being applied to the FIR filter, and to convert the digital signal output from the FIR filter to an analog signal (these converters are not shown in the figures as is the case in the following descriptions).

The impulse response hLL(t) and hLR(t) are obtained in the above mentioned manner, and the sound image is localized on the left side or left rear by using the operational circuits **6** and **7** with a phantom speaker from which the sound is perceived to come.

Similarly, the operational circuits **4** and **5** perform the convolution on the input signals so as to localize the sound image on the right side or right rear.

The output signals from the operational circuits **4** and **5** are applied to the delay circuits **8** and **9** respectively and delayed by τ_1. The output signals from the operational circuits **6** and **7** are applied to the delay circuits **10** and **11** respectively, and delayed by τ_2. An optimal amount of the delay time is about 10 msec. with respect to the input signal, the amount being empirically obtained. An optimal difference between the delay times τ_1 and τ_2 is also experimentally obtained with an amount of about 10 msec. The difference between the delay times τ_1 and τ_2 in the respective phantoms to be localized on the left side and right side allows the phantoms to be distinguished as to whether a phantom is localized on the left side or the right side.

In the next step, the output signals from the delay circuits **8** and **10** are applied to the adder **12**, added to the signal MR(t) input from the input terminal **2**, and mixed with the signal MR(t) at a desirable ratio by the adder **12**, Similarly, the output signals from the delay circuits **9** and **11** are applied to the adder **13**, added to and mixed with the signal ML(t) input from the input terminal **1** at an desirable ratio. The resulting signals are acoustically reproduced by the speakers **14** and **15** respectively.

<u>Example 2</u>

A sound field controller according to a second example of the present invention will be explained with reference to Figure **4**. Figure **4** shows a block diagram of the structure of a sound field controller according to the second example. Circuits having the same functions as the corresponding parts of the sound controller in the first example are represented by the same reference numerals and will not be described in detail.

In Figure **4**, the signals ML(t) and MR(t) applied to the respective input terminals **1** and **2**. These signals are divided into two branches respectively. One of the branched signals of ML(t) and one of the branched signals of MR(t) are applied to a difference signal extractor **3** and the others to adders **13** and **12** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto, and outputs the difference signal to operational circuits **6** and **7**.

Each of the output signals of the operational circuits **6** and **7** is divided into two branches. Two output signals of the operational circuit **6** are applied to the delay circuits **9** and **10**, and two output signals of the operational circuit **7** is applied to the delay circuits **8** and **11**. The output signals from the delay circuits **8** and **10** are applied to the adder **12**, while the output signals from the delay circuits **9** and **11** are applied to the adder **13**.

The delay circuits **8** and **9** delay the input signals by the delay time τ_2, and the delay circuits **10** and **11** delay the input signals by the delay time τ_1. The adder **12** adds the input signal MR(t) from the input terminal **2**, and the output signals from the delay circuits **8** and **10** at an arbitrary ratio. The adder **13** adds the input signal ML(t) from the input terminal **1**, and the output signals from the delay circuits **9** and **11** at an arbitrary ratio. The output signals of the adders **12** and **13** are applied to and produced from speakers **14** and **15** respectively.

In this example, the sound field controller comprises only two operational circuits, each of the output signals form the operational circuits being applied to two delay circuits.

By setting the two impulse responses hLL(t) and hLR(t) inversely in the respective signals which are to be reproduced from the speakers **14** and **15**, the sound image can be localized rightward or leftward in simple manner. For example, to localize the sound image at the right side with respect to the listener, the signals delayed by τ_2 via the delay circuit **8** and **9** are applied crosswise to the adders **12** and **13**. These are two same signals which were used for localizing the sound image at left side.

The above-mentioned configuration is based on the assumption that the impulse responses at the left and right ears of the listener are laterally symmetric. As a result, it is possible to reduce the size of the operational circuits for localizing the left and right sound images by applying one branched signal of the operational circuit straight to the corresponding adder and the other crosswise to the other adder as shown in Figure **4**.

Example 3

A sound field controller according to a third example of the invention will be explained with reference to Figure **5**. Figure **5** shows a block diagram of the structure of a sound field controller according to the third embodiment. Circuits having the same functions as the corresponding parts of the sound field controller in the first and second examples are represented by the same reference numerals and will not be described in detail.

In Figure **5**, the signals ML(t) and MR(t) are applied to the respective input terminals **1** and **2**. These signals are divided into three branches respectively. One of the branched signals of ML(t) and one of the branched signals of MR(t) are applied to a difference signal extractor **3** and converted into the difference signal S(t), and the resulting signal S(t) is applied to delay circuits **19-1** and **19-2**. The delay circuits **19-1** and **19-2** delay the difference signal S(t) by the delay times τ_2 and τ_1 respectively. The other branched signals of ML(t) and MR(t), are applied to a signal judging circuit **20** and a correlator **21**.

The signal judging circuit **20** detects a blank period (i.e. a silent interval where the signal is essentially zero) of the input signal, and judges whether the input signal is a voice signal or non-voice signal. The correlator **21**, on the other hand, is a circuitry for determining the correlation ratio between input signals MR(t) and ML(t). An output signal $S(t-\tau_1)$ from the delay circuit **19-2**, and a output signal $S(t-\tau_2)$ from the delay circuit **19-1** are applied to adders **23** and **22** respectively. The adders **23** and **22** add the input signals thereto with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The resulting signals MR'(t) and ML'(t) are produced from the speakers **14** and **15** respectively.

The operation of the sound field controller according to the third example will be described as to the different portions from the previous examples.

The signal judging circuit **20** adds the input signals MR(t) and ML(t) to obtain a sum signal, detects the frequency of the blank periods (i.e. how frequently the signal interruptions occur) in the sum signal, and judges whether the input signal is a voice signal or not according to the frequency of the blank periods.

Figure **6** shows a waveform of the voice signal. In Figure **6**, the horizontal axis of the coordinate represents the time and the vertical axis of the coordinate represents the amplitude. This sound wave was obtained from the spoken words "DOMO ARIGATO GOZAIMASITA (Thank you very much)" in Japanese as indicated over the waveform. As can seen known from Figure **6**, there will always be a certain number of blanks (silent periods) within a certain period of time in a voice signal (in this example there are two blanks in a 1 second period). The signal judging circuit **20** uses this property of the voice signal to determine whether the input signal is a voice signal or a non-voice audio signal based on the blank period frequency, and controls the summation ratio of the adders **22** and **23**.

A judging value A is set as follows:

for a non-voice audio signal $A = (A + \Delta A)$

for a voice signal $A = (A - \Delta A)$

where Δ A is a constant for varying the amount of the judging value according to whether the signal is a voice signal or not.

When the input signal is determined to be a non-voice audio signal, the judging value A is increased by the constant Δ A, while when the input signal is determined to be a voice signal; the judging value A is decreased by the constant Δ A. This operation is successively repeated at a predetermined interval and the judging value A is updated at each judgment. In this manner, the input signal is judged by variation Δ A of the judging value A from a previously judged value, and not judged by the values **0** or **1** for each judgment. This updating method allows the sound-field controller to handle judging error to prevent any significant effect on the output signals. The judging value A thus determined is applied to the adders **22** and **23**.

The correlator **21** calculates the correlation ratio between the input signals according to following Equation (28) as described below.

$$\alpha = \frac{|ML(t) - MR(t)|}{|ML(t) + MR(t)|} \tag{28}$$

In the case where the input 2ch signals are a monaural signal or an approximately monaural signal (i.e. the 2ch signals MR(t) and ML(t) are strongly correlated each other), the nominator of the equation is zero or decreases to zero, and the value αA becomes nearly zero. When the input 2ch signals are a stereo signal (i.e. the 2ch signals MR(t) and ML(t) have no or little correlation each other), the nominator increases.

The summation ratio of the signals in the adders **22** and **23** is controlled based on the values obtained by the signal judging circuit **20** and the correlator **21**.

The adders **22** and **23** perform summation expressed in the following equations:

$$MR'(t) = MR(t) \cdot (1 - \alpha \cdot A) + S(t - \tau_2) \cdot \alpha \cdot A \tag{29}$$

$$ML'(t) = ML(t) \cdot (1 - \alpha \cdot A) + S(t - \tau_1) \cdot \alpha \cdot A \tag{30}$$

where MR'(t) and ML'(t) are output signals from the adders **22** and **23**, respectively. In these equations, the summing ratios of ML(t), MR(t), and the respective surround signal $S(t - \tau_1)$ and $S(t - \tau_2)$ are adjusted to produce a natural presence. In other words, the correlation ratio between the input signals is small (i.e. giving a listener a large stereophonic feeling), the signal processed by the difference signal extractor **3** is reproduced large, while when the correlation ratio between the input signals is large (i.e. giving a listener a small stereophonic feeling), the signal processed by the difference signal extractor **3** is reproduced small. Further, the voice signal may be reproduced clearly since the judgement of the input signal to be a voice signal or not is performed at the same time and the summation ratio is adjusted.

Although a given by Equation (28) is used with a direct form in Equations (29) and (30), in practice, the value α may be covered into a value in a range of 0 to 1. Further, this value may be varied depending on a desirable magnitude of the stereophonic effects.

In this example, ML(t) and MR(t) are multiplied by a factor $(1 - \alpha \cdot A)$ in order to suppress the change in the total volume of ML'(t) and MR'(t) according to the change of the value α. However, when the total volume is allowed to change, the input signal is not required to be multiplied by $(1 - \alpha \cdot A)$.

The value $\alpha \cdot A$ is updated at a timing with certain time intervals, since the updating operation may cause a fluctuation in the effect.

The value α indicating the correlation ratio may be used in another form of correlation value instead of the exact form. Similarly to the voice judging value A, the correlation value B may be defined as:

$$\text{when } \alpha > X,\ B = (B + \Delta B)$$

$$\text{when } \alpha < X,\ B = (B - \Delta B)$$

where X is a predetermined value and ΔB a constant for varying the correlation value B. The operation using this correlation value is also able to prevent the output signals from fluctuations caused by the updating timing of αA or an erroneous judgment.

According to this example, the input signal is judged to be a voice signal or a non-voice signal by the signal judging circuit **20** based on the frequency of the blank periods. Alternatively, other methods may be used for judgment such as a determining method base on the inclination of the envelope of a rising edge or falling edge of the input signal waveform, or a combination of this determining method with the method in this example.

In this example, the sum signal of the input signals is judged by the signal judging circuit **20**. Alternatively, each

input signal may be judged without summation.

Example 4

A sound field controller according to the fourth example of the invention will be explained with reference to Figure 7. Figure 7 shows a block diagram of the structure of a sound field controller according to the fourth example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

In Figure 7, an Lch signal ML(t) is applied to an input terminal **1** and an Rch signals MR(t) is applied to an input terminal **2**. These signals are divided into branches respectively. One of the branched signals of ML(t) and ore of the branched signal of MR(t) are applied to a difference signal extractor **3** and the others to adders **22-1** and **23-1** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto, and outputs the difference signal to operational circuits **4, 5, 6,** and **7.**

The other branched signals of ML(t) and MR(t) are applied to a signal judging circuit **20** and a correlator **21.**

The signal judging circuit **20** detects any blank period of the input signal, and judges whether the input signal is a voice signal or a non-voice signal. The correlator **21**, on the other hand, is a circuit for determining the correlation ratio between input signals MR(t) and ML(t).

The respective output signals S1(t), S2(t), S3(t), and S4(t) of the operational circuits **4, 5, 6,** and **7** are applied to the adders **22-1** and **23-1** via the delay circuits **8, 9, 10,** and **11.**

The adder **22-1** weights and adds the input signals from the input terminal **2**, the delay circuit **8**, and the delay circuit **10** with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21.** The adder **23-1** weights and adds the input signals from the input terminal **1**, the delay circuit **9**, and the delay circuit **11** with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21.** The output signals MR1'(t) and ML1'(t) from the adders **22-1** and **23-1** are reproduced from the speakers **14** and **15** respectively.

The operation of the sound field controller according to the forth example will be described as to the different portions from the previous examples.

This example is similar to the first example except for the signal judging circuit **20** and the correlator **21.** And the signal judging circuit **20** and the correlator **21** operate the same way as that of the corresponding components of the third example. The operation of the adders **22-1** and **23-1**, however, is somewhat different from that of the third example.

The adder **22-1** performs the summing operation according to the following equation:

$$MR1'(t) = MR(t) \cdot (1 - \alpha \cdot A) + (S1(t) + S2(t)) \cdot \alpha \cdot A \tag{31}$$

In a similar manner, the adder **23-1** performs summing operation as shown in following equation:

$$ML1'(t) = ML(t) \cdot (1 - \alpha \cdot A) + (S3(t) + S4(t)) \cdot \alpha \cdot A \tag{32}$$

The operations of other circuits are similar to those of the previous examples. Also, in order to simplify the structure of the sound field controller, the circuits other than the signal judging circuit **20**, the correlator **21**, and the adders **22-1** and **23-1** may be modified to the corresponding circuits as described in the second example.

Example 5

A sound field controller according to the fifth example of the invention will be explained with reference to the figures. Figure **8** shows a block diagram of the structure of a sound field controller according to the fifth example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

In Figure **8**, an Lch signal ML(t) is applied to an input terminal **1** and an Rch signal MR(t) is applied to an input terminal **2**. These signals are divided into two branches respectively. One of the branched signals of ML(t) and one of the branched signals of MR(t) are applied to a difference signal extractor **3** and the others to adders **12** and **13** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto. The output signal of the difference signal extractor **3** is supplied to reflected sound generation circuits **24** and **25** which generates a reflection and a reverberation by simulating the sound field in a music hall, etc. The outputs of the reflected sound generation circuit **24** is applied to the operational circuits **4** and **5**. The reflected sound generation circuit **25** is applied to the operational circuits **6** and **7.**

The output signals of the operational circuits **4** and **6** are applied to the adder **12** via the delay circuits **8** and **10** respectively. The output signals of the operational circuits **5** and **7** are applied to the adder **13** via the delay circuits **9**

and 11 respectively. The outputs of the delay circuits **9** and **10** are crosswise applied to the adders **12** and **13**.

The adder **12** adds the input signals from the input terminal **2**, the delay circuit **8**, and the delay circuit **10** with respective ratios, while the adder **13** adds the input signals from the input terminal **1**, the delay circuit **9**, and the delay circuit **11** with respective ratios. The output signals from the adders **12** and **13** are reproduced from the speakers **14** and **15** respectively.

The operation of the sound field controller according to the fifth example will be described as to the different portions from the previous examples.

The difference signal produced from the difference signal extractor **3** is applied to the reflected sound generation circuits **24** and **25**. The reflected sound generation circuits **24** and **25** generate a reflection or a reverberation obtained by simulating the sound field in a music hall, etc.

Figures **9A** and **9B** schematically show a reflection series generated by the reflected sound generation circuits **24** and **25**. The horizontal axis of the coordinate represents the time, and the vertical axis of the coordinate represents the amplitude. These reflection series are determined by measurement in an actual music hall or by simulation utilizing the sound ray method.

Figures **10A** and **10B** show diagrams for explaining the reflected sound generation circuits **24** and **25**. An exemplary structure of the reflected sound generation circuits **24** and **25** is shown in Figure **10A**. In Figure **10A**, the signal is applied to a signal input terminal **54-1** and goes through a serially connected I-1 delay elements **51**. Each of delay elements **51** delays the signal by τ_i (i represents a suffix number as in all the following cases), each of multipliers **52** multiplies the input signal by a value called the tap coefficient indicated by X(i), an adder **53** adds all the signals output from each multiplier (called a tap) **52**, and the added (sum) signal is output via an output terminal **54-2**.

The above-mentioned operation is expressed with digital signals. When analog signals are handled in practice, an A/D converter and a D/A converter are to be provided in order to convert the analog signals to digital signals before being applied to the reflected sound generation circuits **24** and **25**, and to convert the digital signals output from the reflected sound generation circuits **24** and **25** to analog signals (these converters are not shown in the figures).

These reflected sound generation circuits **24** and **25** comprise the delay elements **51** and the tap **52** as described above, similarly to the operational circuits **4**, **5**, **6** and **7** in the first example. In this example, each of the delay elements **51** can delay the input signal by respective values of the delay time τ_i, which may vary in each delay circuit. By setting the delay times τ_i and the tap coefficients X(i) appropriately, a desirable reflection series such as shown in Figures **9A**, **9B**, and **10B** are generated by the reflected sound generation circuits **24** and **25**.

The reflected sound generation circuits **24** and **25** may be implemented by using a dynamic random access memory (DRAM) and a digital signal processor (DSP), or the like. Since the reflected sound generation circuits **24** and **25**, and the operational circuits **4**, **5**, **6**, and **7** are configured in the same manner, the functional characteristics of the reflected sound generation circuits **24** and **25** can be included in those of the operational circuits **4**, **5**, **6**, and **7**. As mentioned above, by adding the reflected sound signal to the difference signal (surround signal), the surround feeling given by the difference signal can be emphasized.

The operations of other circuits are similar to those of the previous examples. Also, to simplify the structure of the sound field controller, the circuits other than the signal judging circuit 20, the correlator **21**, the reflected sound generation circuits **24** and **25** may be modified to the corresponding circuits as described in the second example.

<u>Example 6</u>

A sound field controller according to the sixth example of the invention will be explained with reference to Figure **11**. Figure **11** shows a block diagram of the structure of a sound field controller according to the sixth example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

In Figure **11**, an Lch signal ML(t) is applied to an input terminal **1** and an Rch signal MR(t) is applied to an input terminal **2**. These signals are divided into branches respectively. One of the branched signals of the ML(t) and one of the branched signals of the MR(t) are applied to a difference signal extractor **3** and the others to adders **22-1** and **23-1** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto. The output signal of the difference signal extractor **3** is supplied to reflected sound generation circuits **24** and **25** which generate a reflection and a reverberation by simulating the sound field in a music hall, etc. The output of the reflected sound generation circuit **24** is applied to operational circuits **4** and **5**. The output of the reflected sound generation circuit **25** is applied to operational circuits **6** and **7**.

Other branched signals of the ML(t) and the MR(t) are applied to a signal judging circuit **20** and a correlator **21**.

The signal judging circuit **20** detects a blank period of the input signal, and judges whether the input signal is a voice signal or a non-voice audio signal. The correlator **21**, on the other hand, is a circuit for determining the correlation ratio between input signals MR(t) and ML(t).

The respective output signals S1(t), S2(t), S3(t), and S4(t) of the operational circuits **4**, **5**, **6**, and **7** are applied to

the adders **22-1** and **23-1** via the delay circuits **8, 9, 10,** and **11** respectively.

The adder **22-1** weighs and adds the input signals from the input terminal **2**, the delay circuit **8**, and the delay circuit **10** with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The adder **23-1** weighs and adds the input signals from the input terminal **1**, the delay circuit **9**, and the delay circuit **11** with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The output signals from the adders **22-1** and **23-1** are reproduced from the speakers **14** and **15** respectively.

The operation of the sound field controller according to the sixth example is similar to that of the forth example except for the signals input to the operational circuits **4, 5, 6,** and **7**, each of the signals being a sum signal of the difference signal from the difference signal extractor **3** and the reflected sound signal produced by the reflected sound generation circuit **24** or **25**.

Example 7

A sound field controller according to the seventh example of the invention will be explained with reference to Figure **12**. Figure **12** shows a block diagram of the structure of a sound field controller according to the seventh example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

In Figure **12**, an Lch signal ML(t) is applied to an input terminal **1** and an Rch signal MR(t) is applied to an input terminal **2**. These signals are divided into two branches respectively. One of the branched signals of the ML(t) and one of the branched signals of the MR(t) are applied to a difference signal extractor **3** and the others to adders **12-1** and **13-1** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto. The output signal of the difference signal extractor **3** is supplied to reflected sound generation circuits **24** and **25** which generate a reflection and a reverberation by simulating the sound field in a music hall, etc. The output of the reflected sound generation circuit **24** is applied to the adder **12-1**, and the output of reflected sound generation circuit **25** is applied to the adder **13-1**. The speakers **14** and **15** reproduce the signals output from the adders **12-1** and **13-1** respectively.

The difference signal produced by the difference signal extractor **3** is added with a reflected sound signal by the reflected sound generation circuits **24** and **25**. The adder **12-1** sums the signal applied to the input terminal **2** and the output signal of the reflected sound generation circuit **24**. The sum signal is reproduced by the speaker **14**. In a similar way, the adder **13-1** sums the signal applied to the input terminal **1** and the output signal or the reflected sound generation circuit **25**. The sum signal is reproduced by the speaker **15**.

Example 8

A sound field controller according to the eighth example of the invention will be explained with reference to Figure **13**. Figure **13** shows a block diagram of the structure of a sound field controller according to the eight example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

In Figure **13**, an Lch signal ML(t) is applied to an input terminal **1** and an Rch signal MR(t) is applied to an input terminal **2**. These signals are divided into branches respectively. One of the branched signals of the ML(t) and one of the branched signals of the MR(t) are applied to a difference signal extractor **3** and the others to adders **22-2** and **23-2** respectively. The difference signal extractor **3** calculates the difference between the two signals applied thereto. The output signal of the difference signal extractor **3** is supplied to reflected sound generation circuits **24** and **25** which generate a reflection and a reverberation by simulating the sound field in a music hall, etc. The output signal SSR(t) of the reflected sound generation circuit **24** is applied to the adder **22-2**, and the output signal SSL(t) of the reflected sound generation circuit **25** is applied to the adder **23-2**. The speakers **14** and **15** reproduce the signals MR2'(t) and ML2'(t) output from the adders **22-2** and **23-2** respectively.

Other branched signals from ML(t) and MR(t) are applied to a signal judging circuit **20** and a correlator **21**. The signal judging circuit **20** detects any blank period in the input signal, and judges whether the input signal is a voice signal or a non-voice audio signal. The correlator **21**, on the other hand, is a circuit for determining the correlation ratio between input signals MR(t) and ML(t).

The adder **22-2** weights and adds the input signal MR(t) from the input terminal **2** and the signal SSR(t) from the reflected sound generation circuit **24** with a respective ratio based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The adder **23-2** weights and adds the input signal ML(t) from the input terminal **1** and the signal SSL(t) from the reflected sound generation circuit **25** with a respective ratio based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The output signals MR2'(t) and ML2'(t) from the adders **22-2** and **23-2** are reproduced from the speakers **14** and **15** respectively.

The operation of the sound field controller according to the eighth example will be described as to the different portions from the previous examples. The summation operation is performed according to the equations below in a manner

similar to the third embodiment.

$$MR2'(t) = MR(t) \cdot (1 - \alpha \cdot A) + SSR(t) \cdot \alpha \cdot A \quad (33)$$

$$ML2'(t) = ML(t) \cdot (1 - \alpha \cdot A) + SSL(t) \cdot \alpha \cdot A \quad (34)$$

The sum signal MR2'(t) and ML2'(t) output from the adders **22-2** and **23-2** are applied to the speakers **14** and **15** respectively.

Example 9

A sound field controller according to the ninth example of the invention will be explained with reference to Figure **14.** Figure **14** shows a block diagram of the structure of a sound field controller according to the ninth example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

In Figure **14**, an Lch signal ML(t) is applied to an input terminal **1** and an Rch signal MR(t) is applied to an input terminal **2**. These signals are divided into branches respectively. The branched signals of the ML(t) are applied to the adder **13-2**, an adder **55**, and a multiplier circuit **30**, respectively. The branched signals or MR(t) are applied to the adder **12-2**, the adder **55**, and an adder **56**, respectively. The multiplier circuit 30 multiples the input signal by -1, and the output signal from multiplier circuit **30** is applied to the adder **56**. The adder **56** sums the signal MR(t) applied to the input terminal **2** and the output signal from the multiplier circuit **30.** The adder **55** sums the signal ML(t) applied to the input terminal **1** and the signal MR(t) applied to the input terminal **2.**

The output signal of the adder **55** is supplied to reflected sound generation circuits **26** and **27** which generate a reflection and a reverberation by simulating the sound field in a music hall, etc. The output signal of the adder **56** is supplied to reflected sound generation circuits **28** and **29** which generate a reflection and a reverberation by simulating the sound field in a music hall, etc. The reflected sound generation circuits **26** and **27** add the reflection to the output of the adder **55.** The reflected sound generation circuits **28** and **29** add the reflection to the output of the adder **56.** The outputs of the reflected sound generation circuits **26** and **28** are applied to the adder **12-2,** and the outputs of the reflected sound generation circuits **27** and **29** are applied to the adder **13-2.**

The adder **12-2** adds the input signal MR(t) from the input terminal **2** and the signals from the reflected sound generation circuits **26** and **28.** The adder **13-2** adds the input signal ML(t) from the input terminal **1** and the signals from the reflected sound generation circuits **27** and **29.** The output signals from the adders **12-2** and **13-2** are reproduced by the speakers **14** and **15** respectively.

The operation of the sound field controller according to the ninth example will be described as to the different portions from the previous examples.

The adder **56** adds MR(t) and -ML(t), outputting the resulting signal MR(t) - ML(t). In other words, the multiplier **30** and the adder **56** constitute a difference signal extraction means. The output from the adder **56** is divided into two portions which are applied to the reflected sound generation circuits **28** and **29** respectively. The reflection is added to MR(t) - ML(t) and the resulting signal is applied to the adders **12-2** and **13-2**.

Similarly, the adder **55** adds the signal MR(t) and ML(t) to generate a sum signal MR(t) + ML(t). That is, the adder **55** functions as a sum signal generation means. The output from the adder **55** is divided into two portions, each applied to the reflected sound generation circuits **26** and **27.** The reflection is added to MR(t) + ML(t) and resulting signal is applied to the adders **12-2** and **13-2** respectively. The reflected sound generation circuits **26, 27, 28,** and **29** have a similar function as the reflected sound generation circuits **24** and **25** described in the fifth example.

By providing the reflected sound generation circuits and adding the reflection to the difference signal and/or the sum signal of the input signals as described above, a sound field can be reproduced with natural expansion and natural presence without the antiphase feeling. Convoluting the reflection into the sum signal of the input signals makes the expansion and presence of the reproduced sound field more effective and more natural. Further, providing two reflected sound generation circuits for each channel makes it possible to reproduce a sound field in which the signals produced from the speakers **14** and **15** have different reflections. That is to say, the reflection can be added in stereo. Further, by varying the amount of delay time of the delay circuit or changing the coefficient of the multiplier in the reflected sound generation circuit, various sound fields such as a sound field with plenty of reverberation or that with little amount of reflection can be reproduced.

Example 10

A sound field controller according to the tenth example of the invention will be explained with reference to Figure **15.** Figure **15** shows a block diagram of the structure of a sound field controller according to the tenth example. The

circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

The adder **22-3** weighs and adds the input signal MR(t) from the input terminal **2**, the signal S1'(t) from the operational circuit **26**, and the signal S2'(t) from the operation circuit **28** with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The adder **23-3** weighs and adds the input signal ML(t) from the input terminal **1** and the signal S3'(t) from the operation circuit **27**, and the signal S4'(t) from the operation circuit **29** with a respective ratio based on the calculated result obtained from the signal judging circuit **20** and the correlator **21**. The output signals MR3'(t) and ML3'(t) from the adders **22-3** and **23-3** are reproduced from the speakers **14** and **15** respectively.

The adders **22-3** and **23-3** perform the addition in the same manner as the third example as follows:

$$MR3'(t) = MR(t) \cdot (1 - \alpha \cdot A) + (S1'(t) + S2'(t)) \cdot \alpha \cdot A \qquad (35)$$

$$ML3'(t) = ML(t) \cdot (1 - \alpha \cdot A) + (S3'(t) + S4'(t)) \cdot \alpha \cdot A \qquad (36)$$

Example 11

A sound field controller according to the eleventh example of the invention will be explained with reference to Figure **16**. Figure **16** shows a block diagram of the structure of a sound field controller according to the eleventh example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

As is shown in Figure **16**, the sound field controller according to the eleventh example compared with that of the ninth example, instead of the adders **12-2** and **13-2**, comprises an adder **12-3** for adding the signals from the reflected sound generation circuits **26** and **28**, and an adder **13-3** for adding the signals of the reflected sound generation circuits **27** and **29**. The sound field controller according to the eleventh example further comprises a multiplier circuit **31** for multiplying the input signal by -1, an adder **13-4** for adding the signals from the adder **12-3** and the multiplier circuit **31** to the input signal ML(t), and an adder **12-4** for adding the output signals from the adder **12-3** and the multiplier **31** to the input signal MR(t). In other words, the adder **12-4** produces a difference signal of the output signals from the adders **12-3** and **13-3**, and the adder **13-4** produces a sum signal of output signals from the adders **12-3** and **13-3**. The output signals from the adders **12-4** and **13-4** are reproduced by the speakers **14** and **15** respectively.

The operation of the sound field controller according to the eleventh example will be described as to the different portions from the previous examples.

The adder **56** adds MR(t) and -ML(t), outputting the resulting signal MR(t) - ML(t). In other words, the multiplier **30** and the adder **56** constitute a difference signal extraction means. The output from the adder **56** is divided into two portions which are applied to the reflected sound generation circuits **28** and **29** respectively. The reflection is added to MR(t) - ML(t) and the resulting signal is applied to the adders **12-3** and **13-3**.

Similarly, the adder **55** adds the signal MR(t) and ML(t) to generate a sum signal MR(t) + ML(t). That is, the adder **55** functions as a sum signal generation means. The output from the adder **55** is divided into two portions, each applied to the reflected sound generation circuits **26** and **27**. The reflection is added to MR(t) + ML(t) and the resulting signal is applied to the adders **12-3** and **13-3** respectively.

The reflected sound generation circuits **26, 27, 28,** and **29** have a similar function as the reflected sound generation circuits **24** and **25** described in the fifth example. The output signals from the reflected sound generation circuits **26** and **28** are applied to the adder **12-3**, and the output signals from the reflected sound generation circuits **27** and **29** are applied to the adder **13-3**.

The adder **12-3** adds the outputs of the reflected sound generation circuits **26** and **28**, with the resulting signal being divided into two portions. One of the signals is applied to the multiplier **31** and the other to the adder **13-4**. The adder **13-3** adds the outputs of the reflected sound generation circuits **27** and **29**, with the resulting signal being divided into two portions. One of the signals is applied to the multiplier **31** and the other to the adder **13-4**. The adder **12-4** multiplies the output signal from the adder **13-3** by -1 and applies the resulting signal to the adder **12-4** and the adder **13-4**. The adder **12-4** adds the input signal MR(t), the output of the adder **12-3** and the output from the multiplier **31**, and applies the resulting sum signal to the speaker **14**. In similar manner, the adder **13-4** adds the input signal ML(t), the output of the adder **12-3**, and the output of the adder **13-3**, and applies the resulting signal to the speaker **15**.

In this way, the output signals from the reflected sound generation circuits **26** and **28**, which are produced by the speaker **14**, are in the same phase (i.e. inphase) with each other. On the other hand, the output signals from the reflected sound generation circuits **27** and **29**, which are produced by the speaker **15** are in antiphase each other.

As explained above, the difference signal and the sum signal of the input stereo signals MR(t) and ML(t) are divided into two portions respectively. One portion of the difference signal and one portion the sum signal are reproduced in the same-phase, and the other portion of the difference signal and the other portion of the sum signal are reproduced in

antiphases each other. Consequently, the feeling of expansion is obtained by antiphase reproduction, and at the same time, any uncomfortable antiphase feeling is attenuated by adding the same-phased signals to the antiphased signals to be reproduced.

Example 12

A sound field controller according to the twelfth example of the invention will be explained with reference to the Figure **17**. Figure **17** shows a block diagram of the structure of a sound field controller according to the twelfth example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

As is shown in Figure **17**, the sound field controller according to the twelfth example, compared with that of the eleventh example, further comprises a signal judging circuit **20** and a correlator **21**, and comprises an adder **22-4** for weighting and adding the signals with respective ratios based on the calculated result obtained from the signal judging circuit **20** and the correlator **21** instead of the adder **12-4**, and an adder **23-4** instead of the adder **13-4**.

The operation of the sound field controller according to the twelfth example will be described as to the different portions from the previous examples.

The adder **22-4** is supplied with the signal SS1(t) output from the adder **12-3**, the signal SS2(t) output from the multiplier **31**, and the input signal MR(t) from the input terminal **2**. The adder **23-4**, on the other hand, is supplied with the signal SS3(t) output from the adder **12-3**, the signal SS4(t) output from the adder **13-3**, and the input signal ML(t) applied to the input terminal **1**. The adders **22-4** and **23-4** perform summation according to the equations as shown below in a manner similar to the third example.

$$MR4'(t) = MR(t) \cdot (1 - \alpha \cdot A) + (SS1(t) + SS2(t)) \cdot \alpha \cdot A \tag{37}$$

$$ML4'(t) = ML(t) \cdot (1 - \alpha \cdot A) + (SS3(t) + SS4(t)) \cdot \alpha \cdot A \tag{38}$$

The output signals MR4'(t) and ML4'(t) from the adders **22-4** and **23-4** are thus produced by the speakers **14** and **15**.

Example 13

A sound field controller according to the thirteenth example of the invention will be explained with reference to the figures. Figure **18** shows a block diagram of the structure of a sound field controller according to the thirteenth example. The circuits having the same functions as the corresponding parts of the sound field controller in the previous examples are represented by the same reference numerals and will not be described in detail.

The signal ML(t) to be reproduced from an Lch and the signal MR(t) to be reproduced from an Rch as viewed from the listener **16** are applied to the input terminals **1** and **2** respectively. Each of these signals is divided into two branches. The branched signals of ML(t) are applied to the reflected sound generation circuits **57** and **58**, and those of MR(t) to the reflected sound generation circuits **59** and **60**. The reflected sound generation circuits **57**, **58**, **59**, and **60** generate a reflection and a reverberation by simulating the sound field in a music hall, etc.

The output signal from the reflected sound generation circuits **57** and **60** are applied to the adders **12-4** and **13-4** respectively. The output signal from the reflected sound generation circuit **58** is further divided into two branch signals and applied to the operational circuits **4** and **5**, and the output signal from the reflected sound generation circuit **59** is divided into two branch signals and applied to the operational circuits **6** and **7**. These operational circuits digitally process the head related transfer function in a time domain in such a manner as to localize the sound on the left and right sides or left and right rear of the listener **16**.

The output signals of the operational circuits **4** and **6** are applied to the adder **12-4** and the output signals of the operational circuits **5** and **7** are applied to the adder **13-4**. The adders **12-4** and **13-4** are also supplied with the output signals from the reflected sound generation circuits **57** and **60**, and output sum signals to the speakers **14** and **15** respectively.

The operation of the sound field controller according to this example will be explained with reference to Figures **18**, and **19A** to **19C**.

The 2ch signals ML(t) and MR(t) are applied to the input terminals **1** and **2**, and then to the reflected sound generation circuits **57** and **58**, and **59** and **60**, respectively. The reflection and/or reverberation is generated by the reflected sound generation circuits **57** and **58** functioning as a pair, and by the reflected sound generation circuits **59** and **60** as another pair.

Figures **19A** and **19B** show a reflection series generated by the reflected sound generation circuits **57** and **58** schematically. In Figures **19A** and **19B**, the horizontal axis of the coordinate represents the time, and the vertical axis of the

coordinate represents the amplitude. For example, when the output signal from the reflected sound generation circuit **58** is localized on the right side or right rear other than the position of the speaker **14** or **15** by using the operational circuits **4** and **5**, the delay time and the amplitude of the reflection in the reflected sound generation circuits **57** and **58** are set up as shown in Figures **19A** and **19B** respectively.

Assuming that the output signal of the reflected sound generation circuit **58** can be processed and played electrically (or virtually) at the position of the speaker **61** as shown in Figure **19C**, and when the delay time and amplitudes of the reflection generated by the reflected sound generation circuits **57** and **58** are set up as shown in Figures **19A** and **19B**, the output signal of the reflected sound generation circuit **58** is perceived to be produced from the speaker **61** and the output signal of the reflected sound generation circuit **57** is produced from the speaker **14.** The components of the reflection are indicated by the letters A to E in the Figures **19A** to **19C.**

In this reproduction process, a sound image is perceived to be synthesized by the human aural characteristics, and recognized as if the reflection is coming from the positions between the speakers **14** and **61** shown in Figure **19C** (See "Spatial Acoustics" by Jens Blauert et al., Kajima Publishing Co., Ltd.). In Figure **19C**, the reflection is indicated by vectors with each length corresponding to the magnitude of the sound (component). Also, the reflections shown in Figures **19A** and **19B** have a time delay. In order to synthesize the reflection between the speakers **14** and **61**, the time difference between the reflections from the two speakers may be used as well as the amplitude difference.

These reflections to be produced can be obtained by measurement in an actual hall or by simulation utilizing the sound ray method or the like. The reflected sound generation circuits **57, 58, 59,** and **60** for generating these reflections have the same structure as the corresponding circuits in the seventh example. Similarly, in the reflected sound generation circuits **59** and **60**, the delay time and the amplitude of reflections are set up such that the reflection is synthesized leftward.

The output signal from the reflected sound generation circuit **58** is divided into two branch signals and applied to the operational circuits **4** and **5** for localizing the sound on the right side or right rear of the listener **16.** Similarly, the output signal from the reflected sound generation circuit **59** is divided into two branch signals and applied to the operational circuits **6** and **7** for localizing the sound on the left side or left rear of the listener **16.** These operational circuits perform a convolution and apply the resulting signals to the corresponding adders respectively. The sum signals from the adders are reproduced by the speakers **14** and **15,** whereby providing (i.e. localizing) a phantom speaker on the left and/or right sides of the listener **16** at the same time. As described above, therefore, the reflections are synthesized and produced between the phantom speaker(s) and the speakers **14** and **15**.

As described above, according to the present invention, a sound field controller is provided in which a reflection and/or a reverberation is generated by adjusting the delay time and the amplitude of reflected sound generation circuits. Further, a sound to be reproduced including the reflection can be perceived to be come from a place other than the reproduction point of the speaker. It is thus possible to reproduce a sound with presence without using any additional speakers on the sides or rear of the listener.

According to the present invention, a sound field controller is provided in which the summation ratio of the surround signal (such as the reverberation and the reflection) and the input stereo signals are appropriately adjusted so as to reproduce a sound with presence retaining a desirable clear sound. In other words, the surround signal is effectively reproduced without making the main signal unclear.

Various other modifications will be apparent to and can be readily made by those skilled in the art without departing from the scope and spirit of this invention. Accordingly, it is not intended that the scope of the claims appended hereto be limited to the description as set forth herein, but rather that the claims be broadly construed.

Claims

1. A sound field controller for reproducing a sound field with presence comprising;

    input means (1, 2) for inputting an input audio signal having two channel signals (MR(t), ML(t)),
    signal extracting means (3) for receiving and processing the input audio signal, and producing an extracted signal from the input audio signal,
    delay means (19-1, 19-2; 8, 9, 10, 11) for delaying the extracted signal by a predetermined time, and producing a delayed signal,
    signal judging means (20) for receiving the input audio signal and judging whether the input audio signal is a voice signal or a non-voice audio signal to output a detecting signal indicating the result,
    correlation determining means (21) for determining correlation ratio between the two channel signals of the input signal to output a determining signal,
    adding means (22, 23; 22-1, 23-1) for receiving the input audio signal, the delayed signal, the detecting signal, and the determining signal, adding the input audio signal and the delayed signal with a predetermined summation ratio based on the detecting signal and the determining signal, and producing a resulting summed signal,

and

output means (14, 15) for reproducing the summed signal.

2. A sound field controller according to claim 1, characterized in that, the controller further comprises

signal processing means (24, 25) for receiving the extracted signal, and for adding a reflected sound signal and/or a reverberated sound signal to the extracted signal to produce a processed signal.

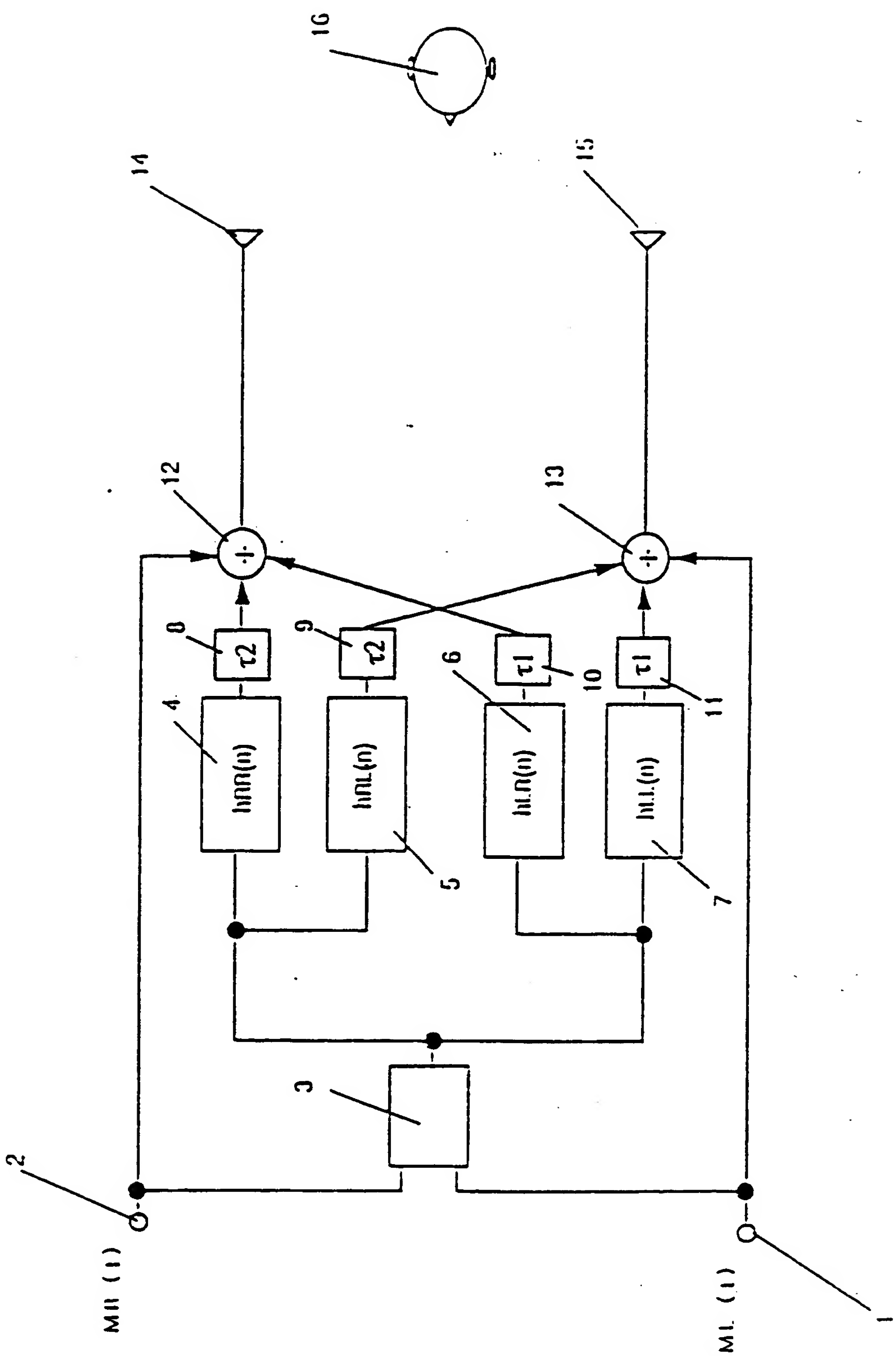
3. A sound field controller according to claim 1 or 2, characterized in that the controller further comprises

operation means for receiving the extracted signal from the signal extracting means (3), performing a convolution on the extracted signal, and generating a convolution sum signal, the operation means comprising a first (4), a second (5), a third (6), and a forth (7) operation portion, wherein
the delay means comprises a first (8), a second (9), a third (10), and a forth (11) delay element, each delay element receiving the convolution sum signal from the corresponding operation portion (5, 6, 7, 8), and
the adding means comprises a first and a second adder, the first adder (22-1) receving the second channel signal (MR(t)) of the input signal and the delayed signal from the first and the third delay element (8, 10), the second adder (23-1) receiving the first channel signal (ML(t)) of the input audio signal and the delayed signal from the second and the forth delay element (9, 11).

4. A sound field controller according to claim 2, characterized in that the controller further comprises

operation means for receiving the processed signal from the signal processing means (24, 25), performing a convolution on the processed signal, and generating a convolution sum signal, the operation means comprising a first (4), a second (5), a third (6), and a forth (7) operation portion, wherein
the delay means comprises a first (8), a second (9), a third (10), and a forth (11) delay element, each delay element receiving the convolution sum signal from the corresponding operation portion (5, 6, 7, 8), and
the adding means comprises a first and a second adder, the first adder (22-1) receving the second channel signal (MR(t)) of the input signal and the delayed signal from the first and the third delay element (8, 10), the second adder (23-1) receiving the first channel signal (ML(t)) of the input audio signal and the delayed signal from the second and the forth delay element (9, 11).

Fig. 1



MR(t)
ML(t)
hRR(n)
hRL(n)
hLR(n)
hLL(n)
τ2
τ2
τ1
τ1
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16

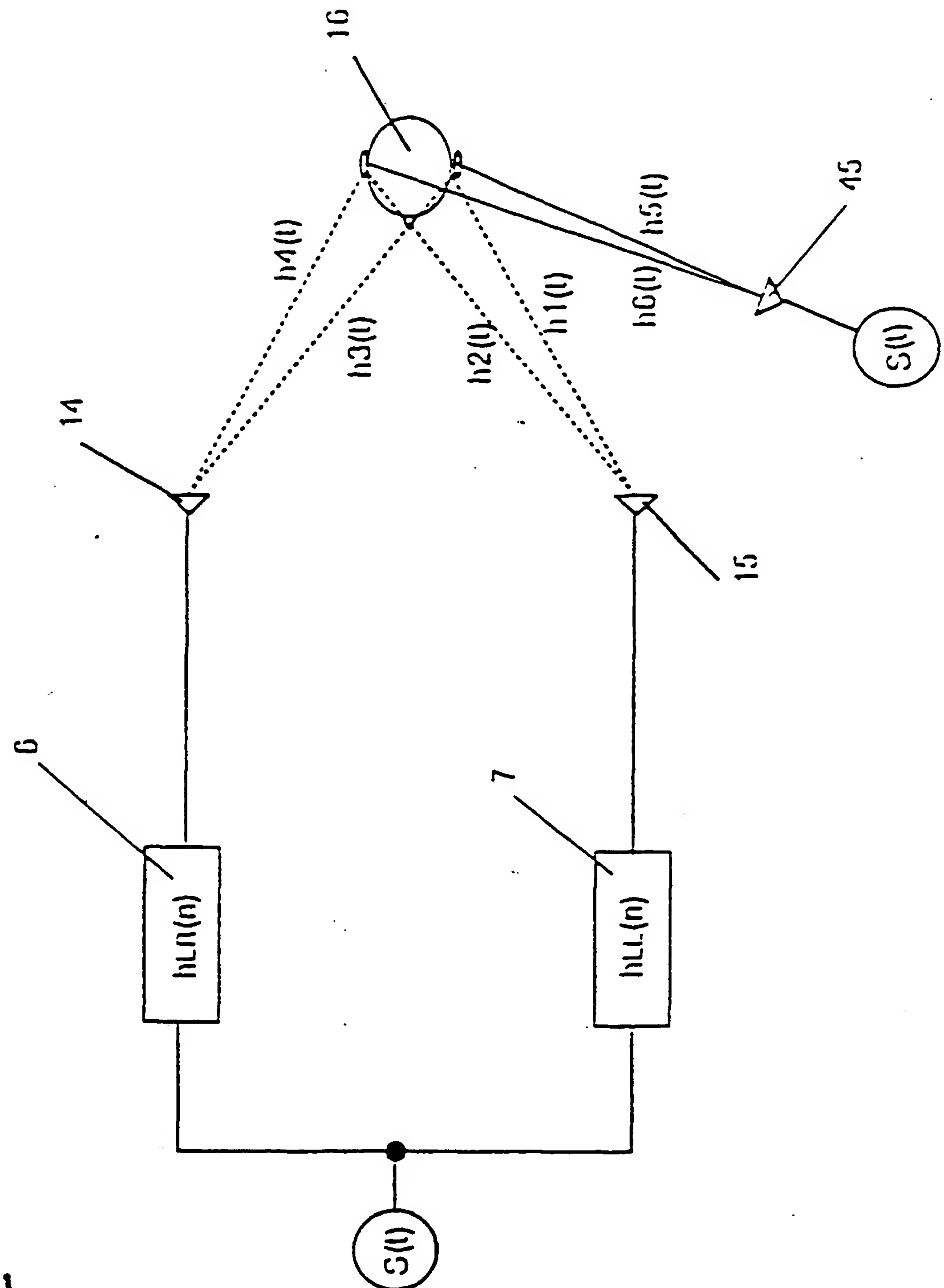


Fig.2

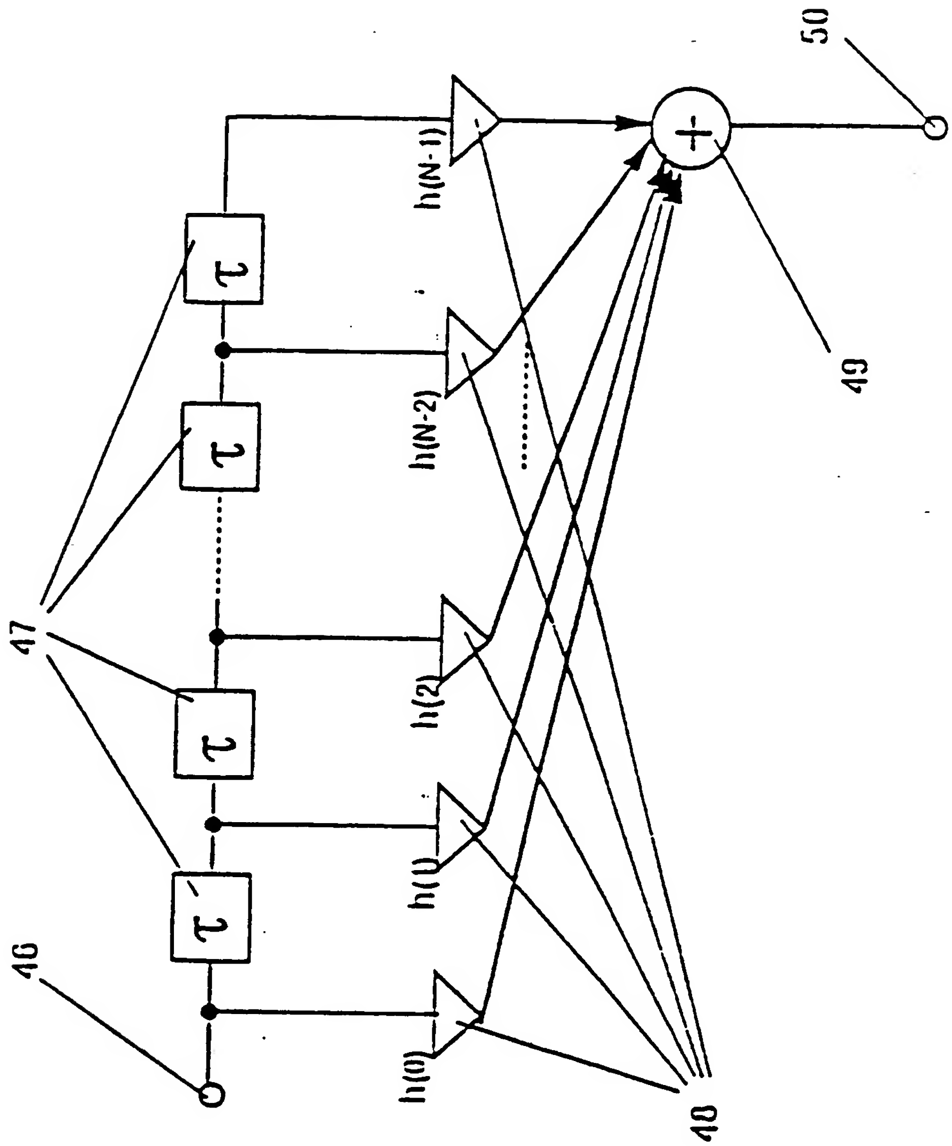


Fig. 3

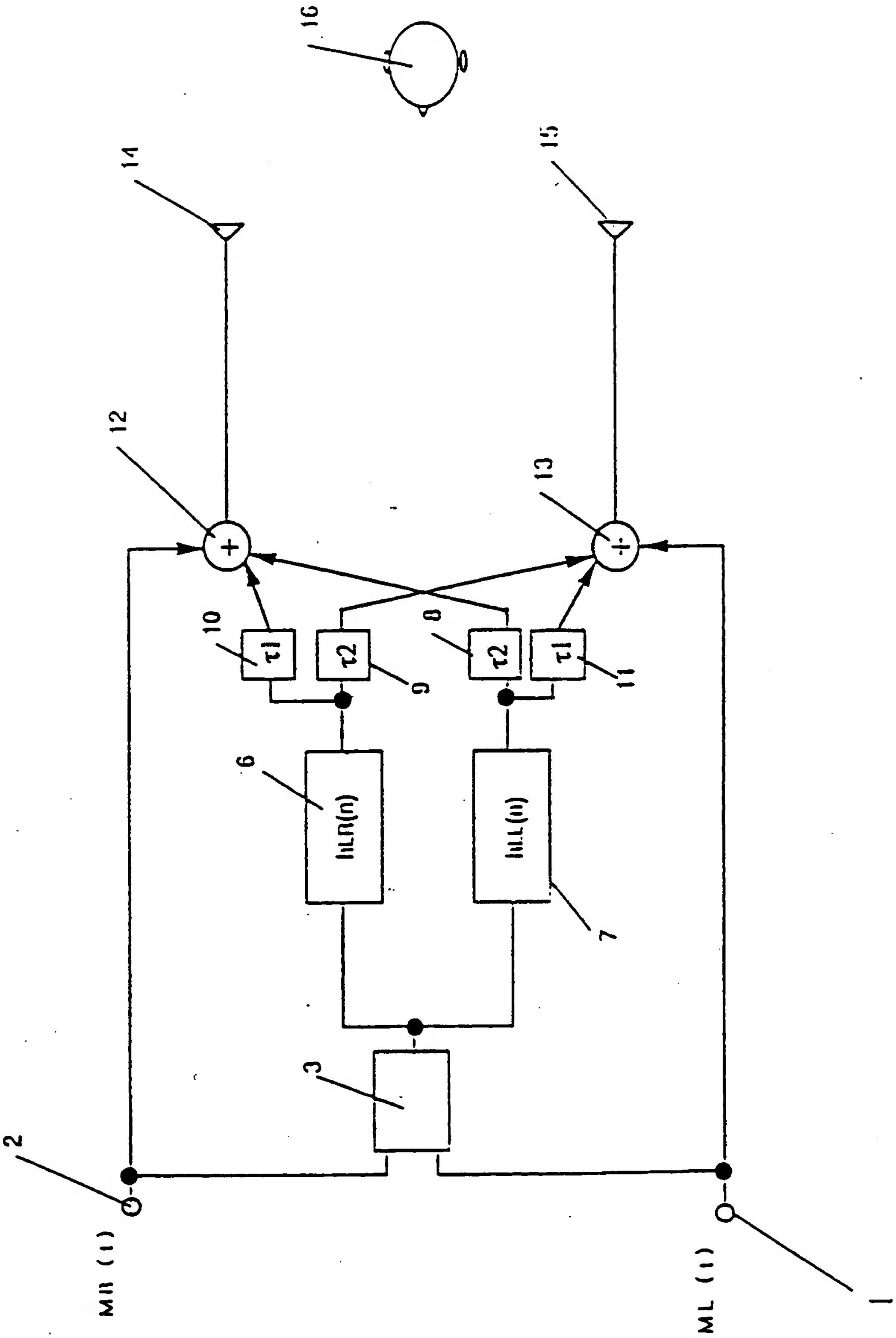
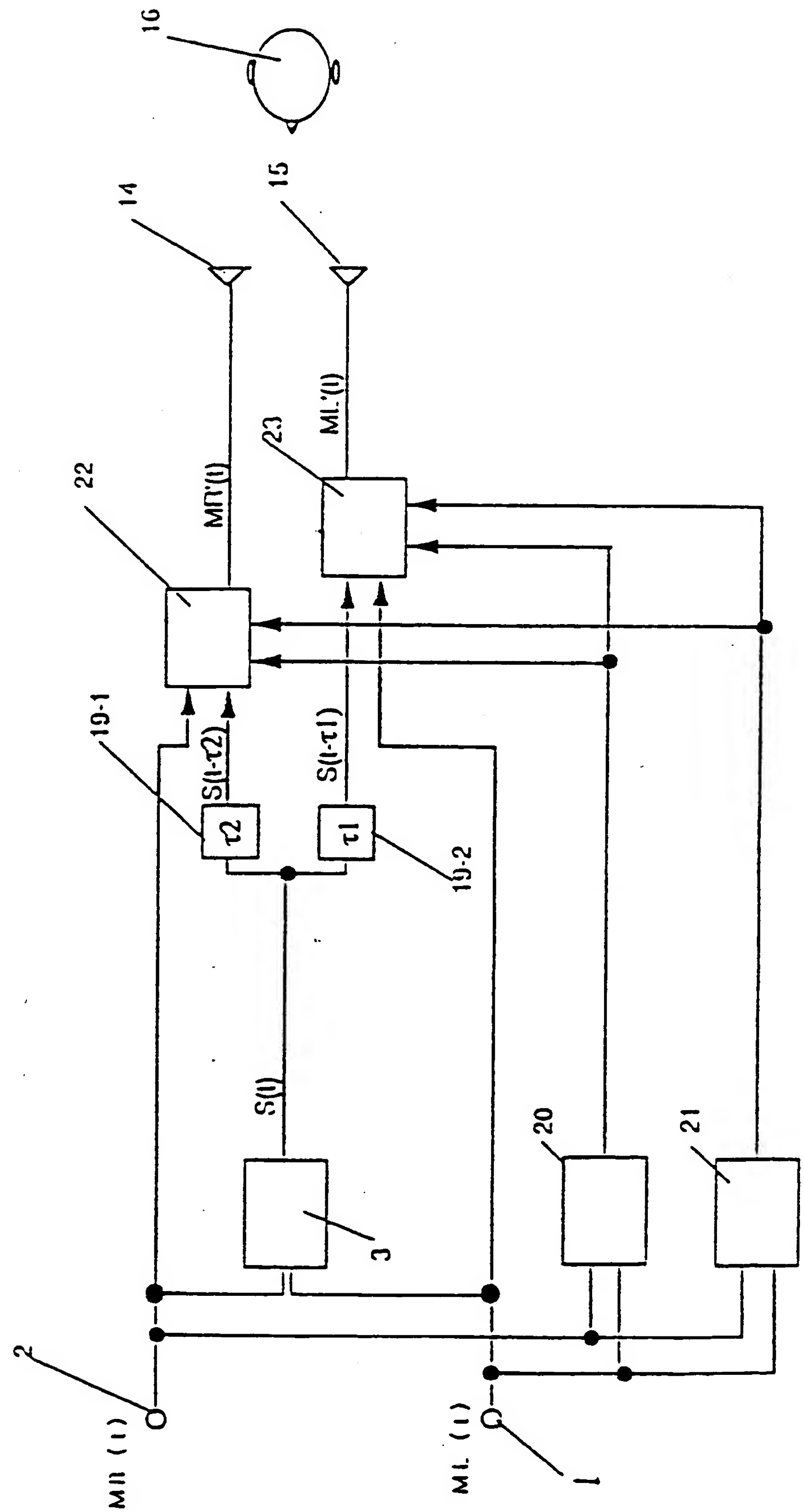


Fig.4

Fig. 5



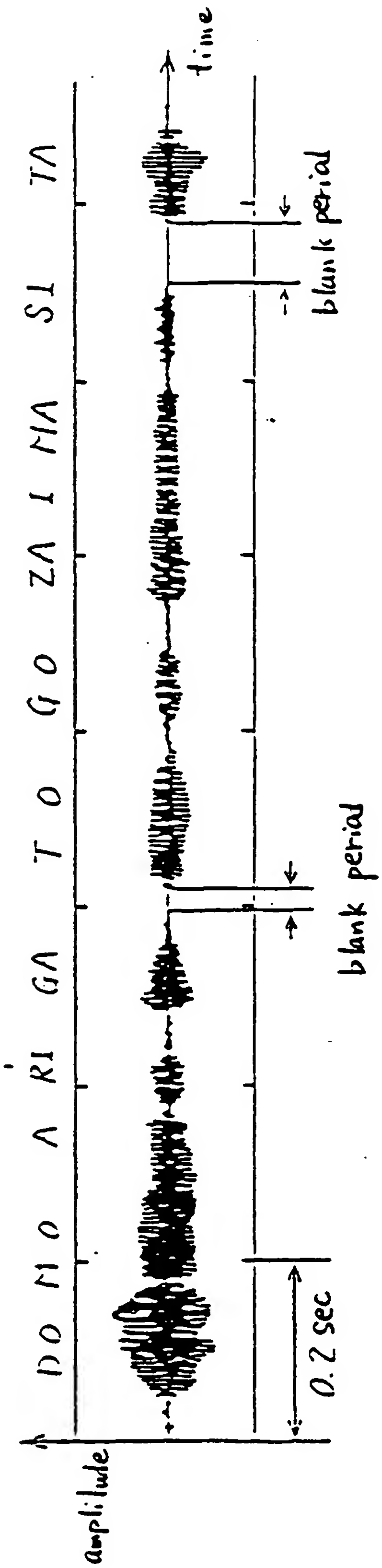

time
amplitude
DO MO A RI GA TO GO ZA I MA SI TA
blank period
blank period
0.2 sec

Fig.6

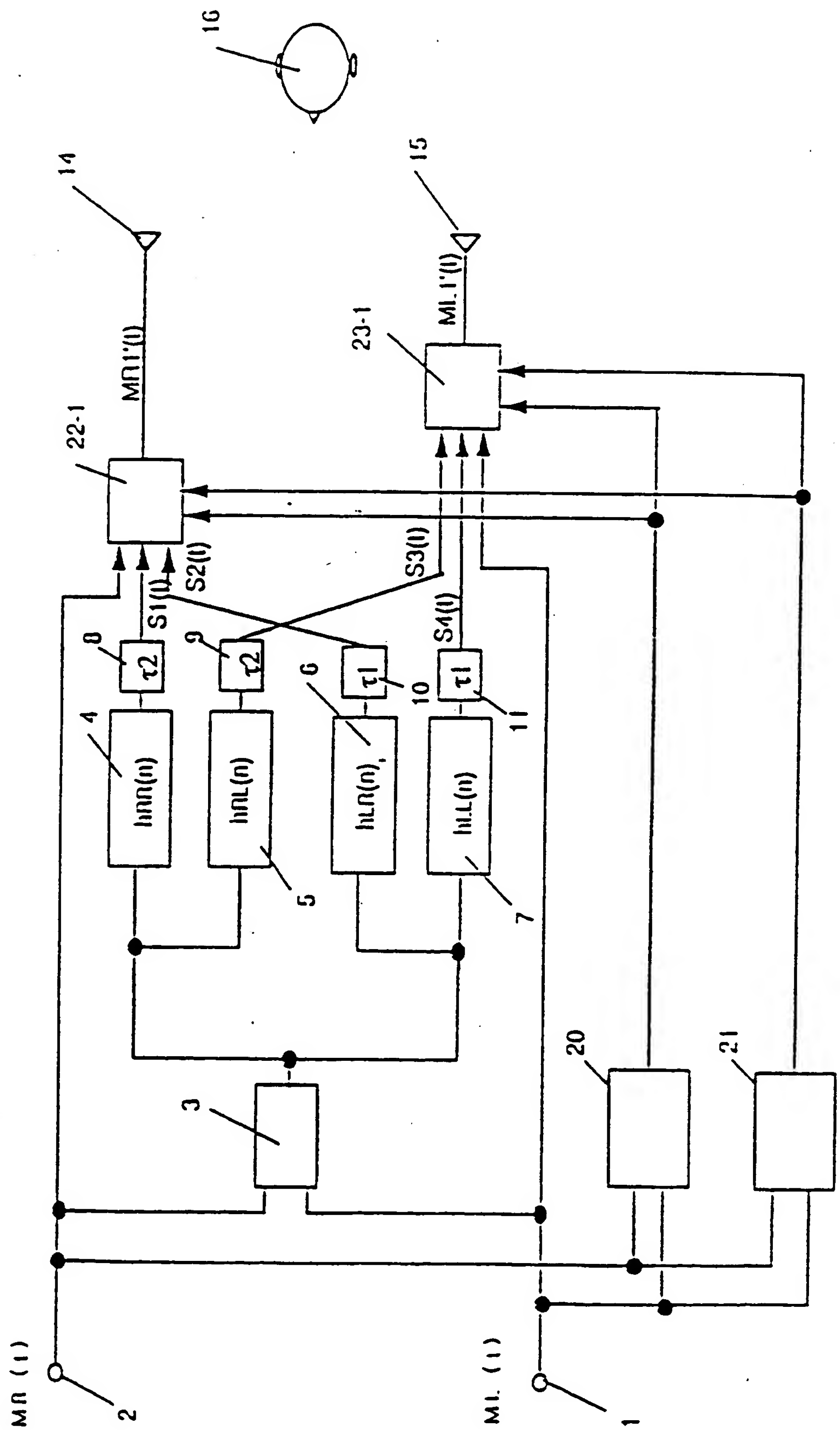

16
14
15
MR1'(t)
ML1'(t)
22-1
23-1
S1(t)
S2(t)
S3(t)
S4(t)
8
9
10
11
τ2
τ2
τ1
τ1
4
5
6
7
hRR(n)
hRL(n)
hLR(n)
hLL(n)
3
20
21
MR(t)
ML(t)
2
1

Fig. 7

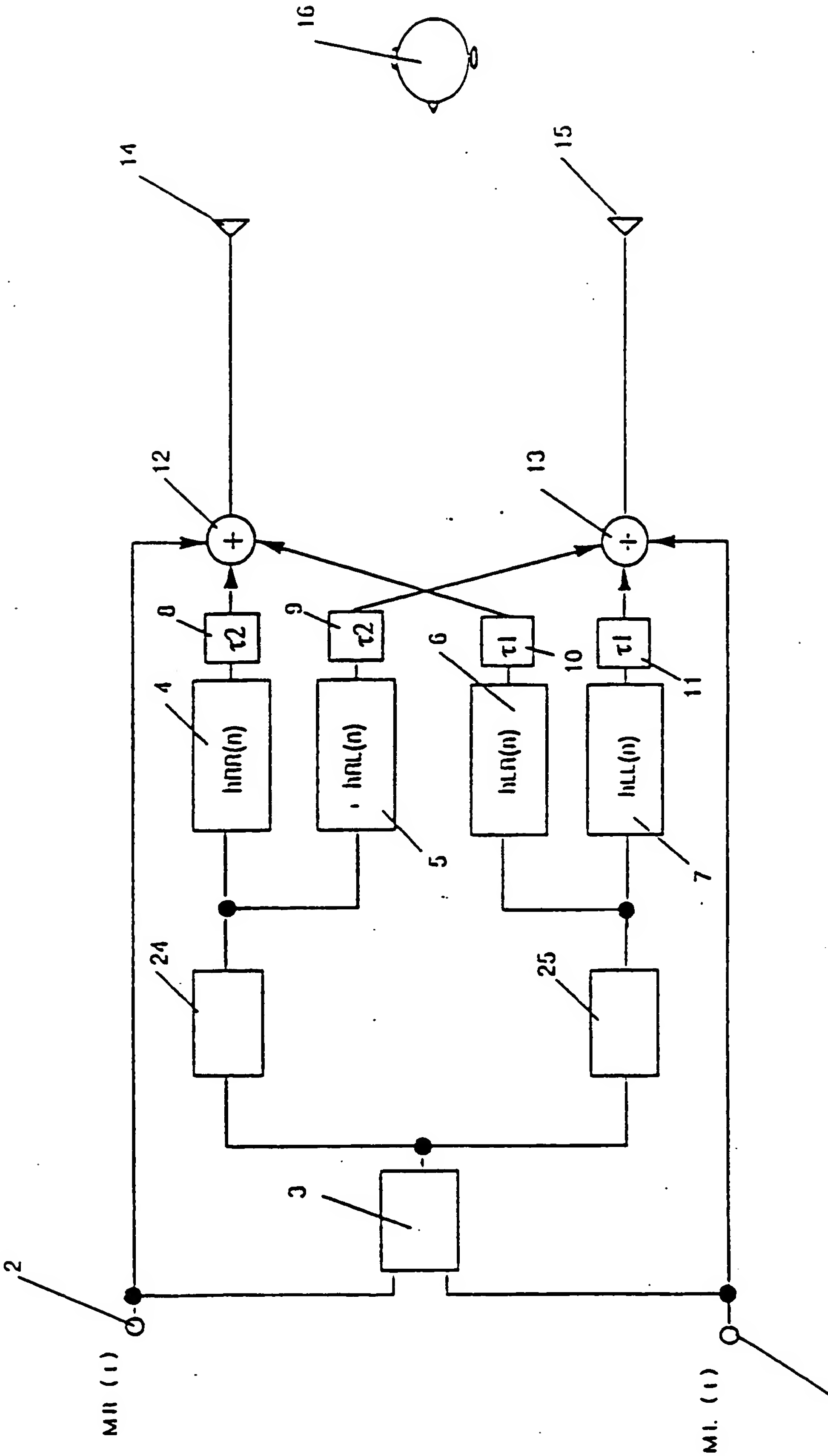


Fig.8

Fig. 9A

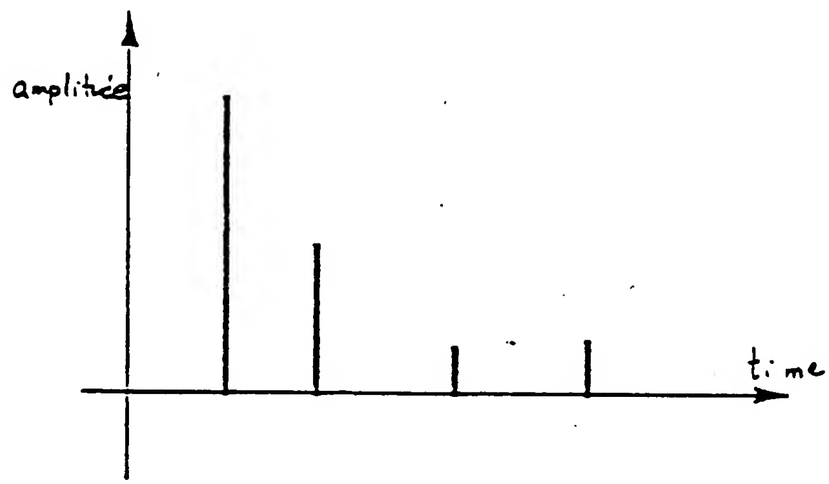
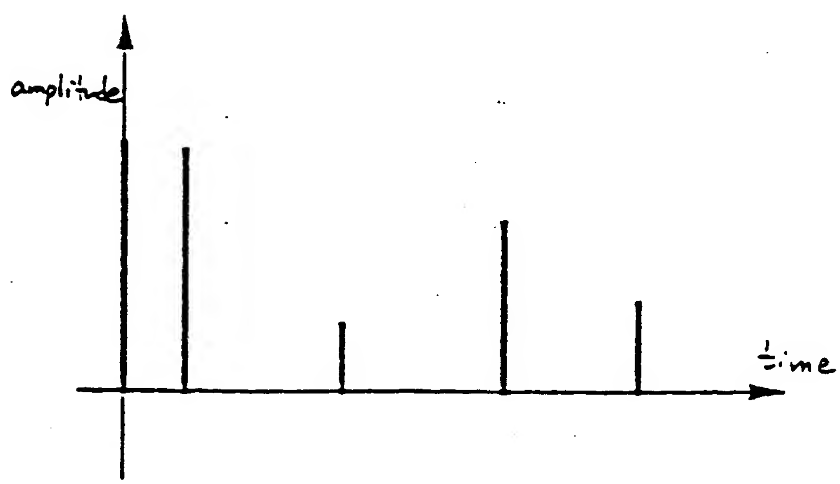


Fig. 9B



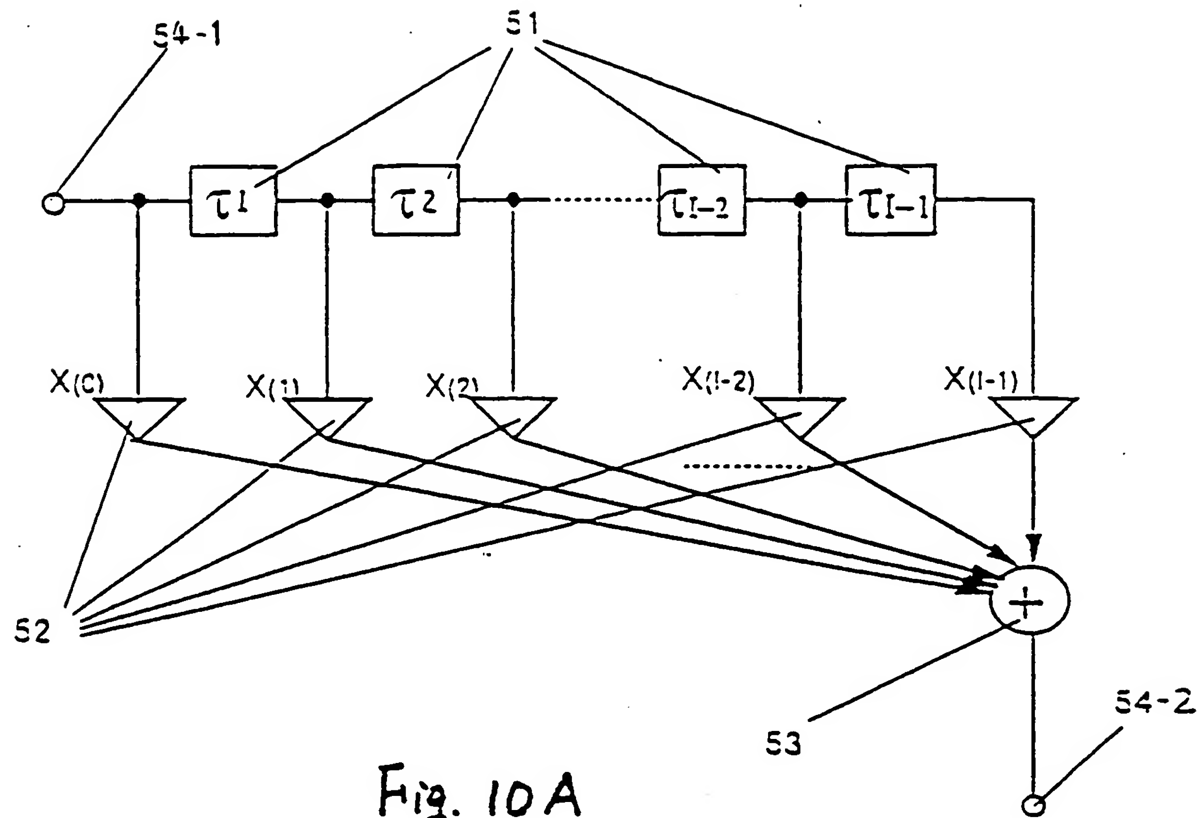


Fig. 10A

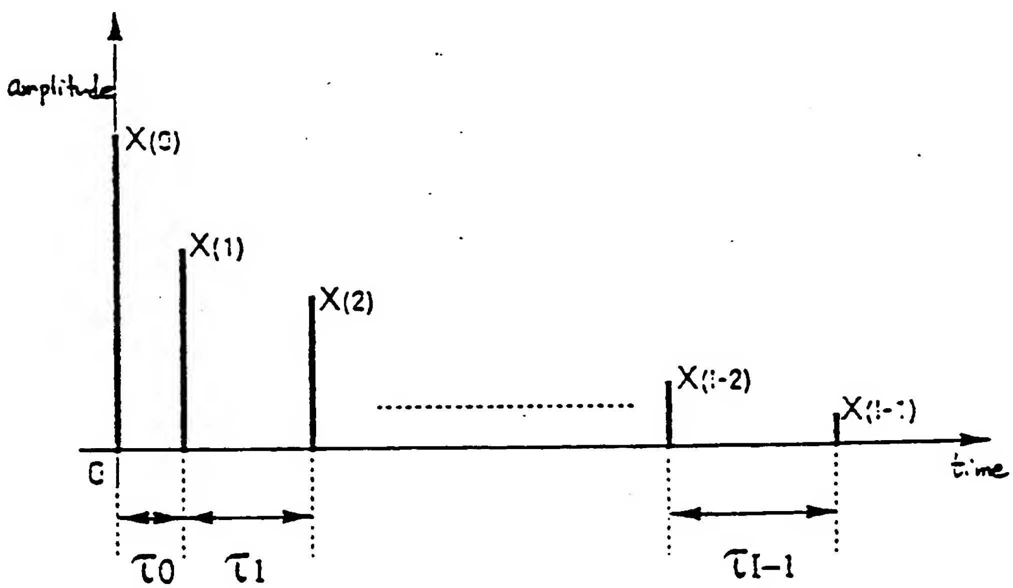


Fig. 10B

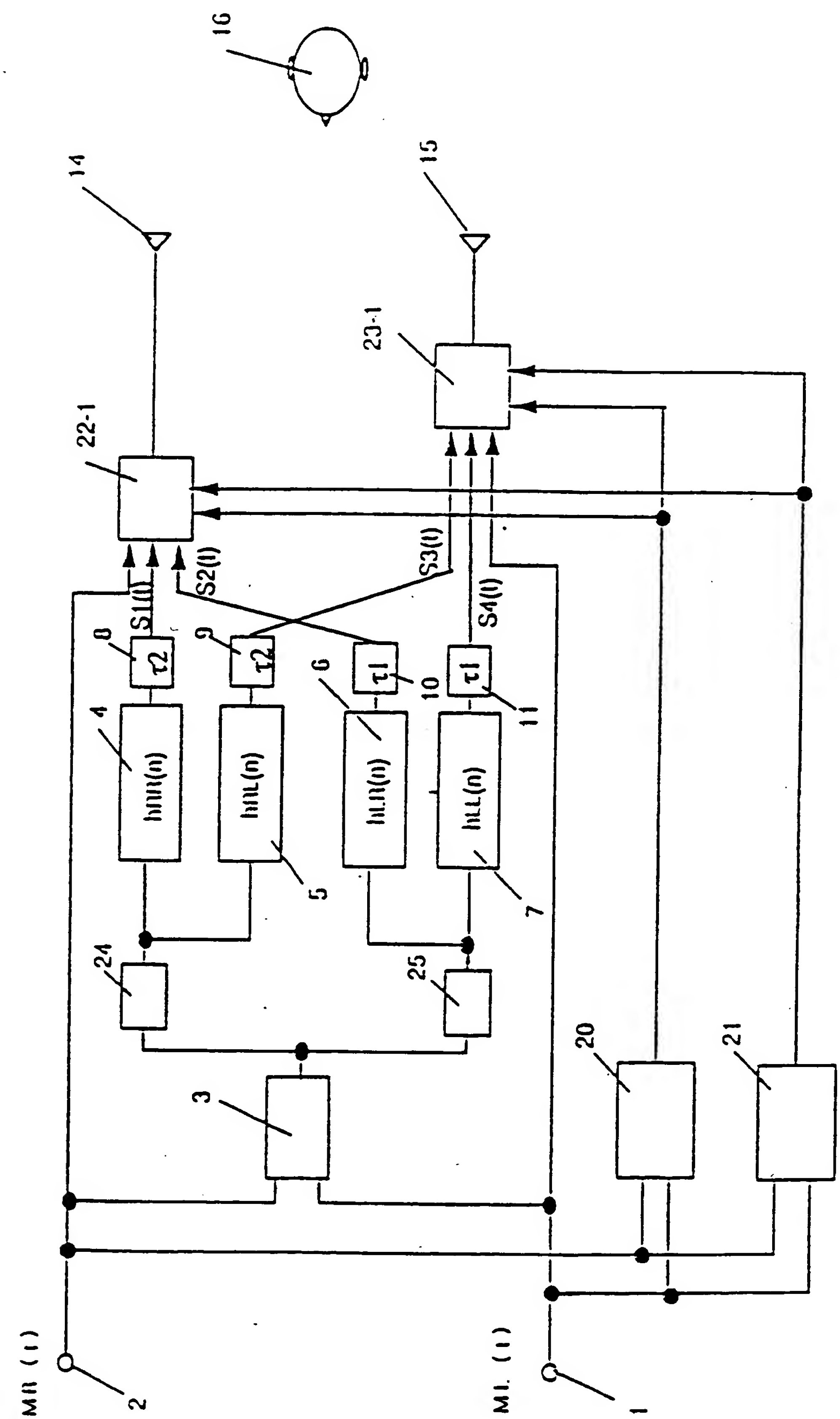
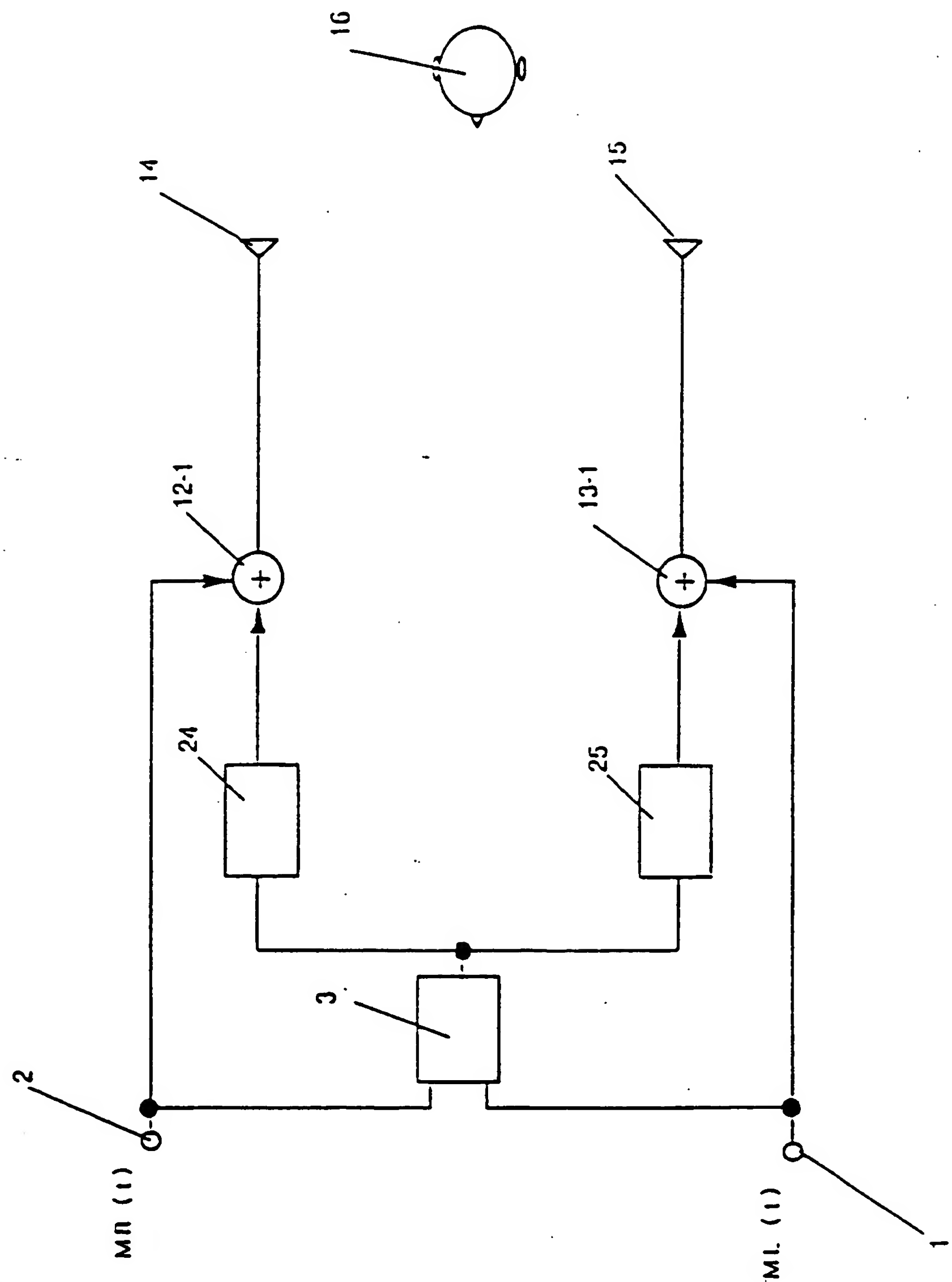

MR (t)
ML (t)
hRR(n)
hRL(n)
hLR(n)
hLL(n)
τ2
τ2
τ1
τ1
S1(t)
S2(t)
S3(t)
S4(t)
22-1
23-1
14
15
16
1
2
3
4
5
6
7
8
9
10
11
20
21
24
25

Fig. 11

Fig. 12



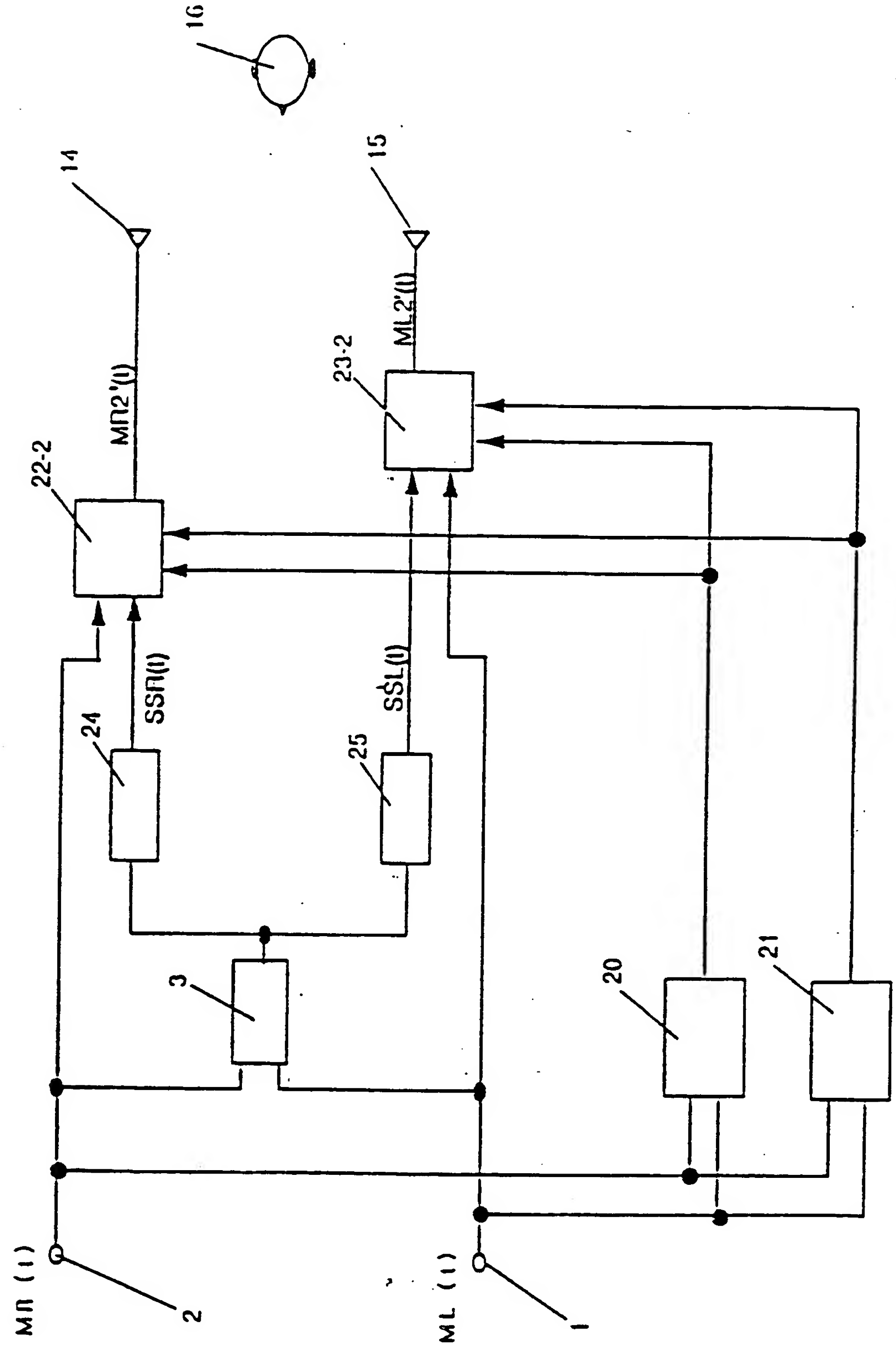


Fig.13

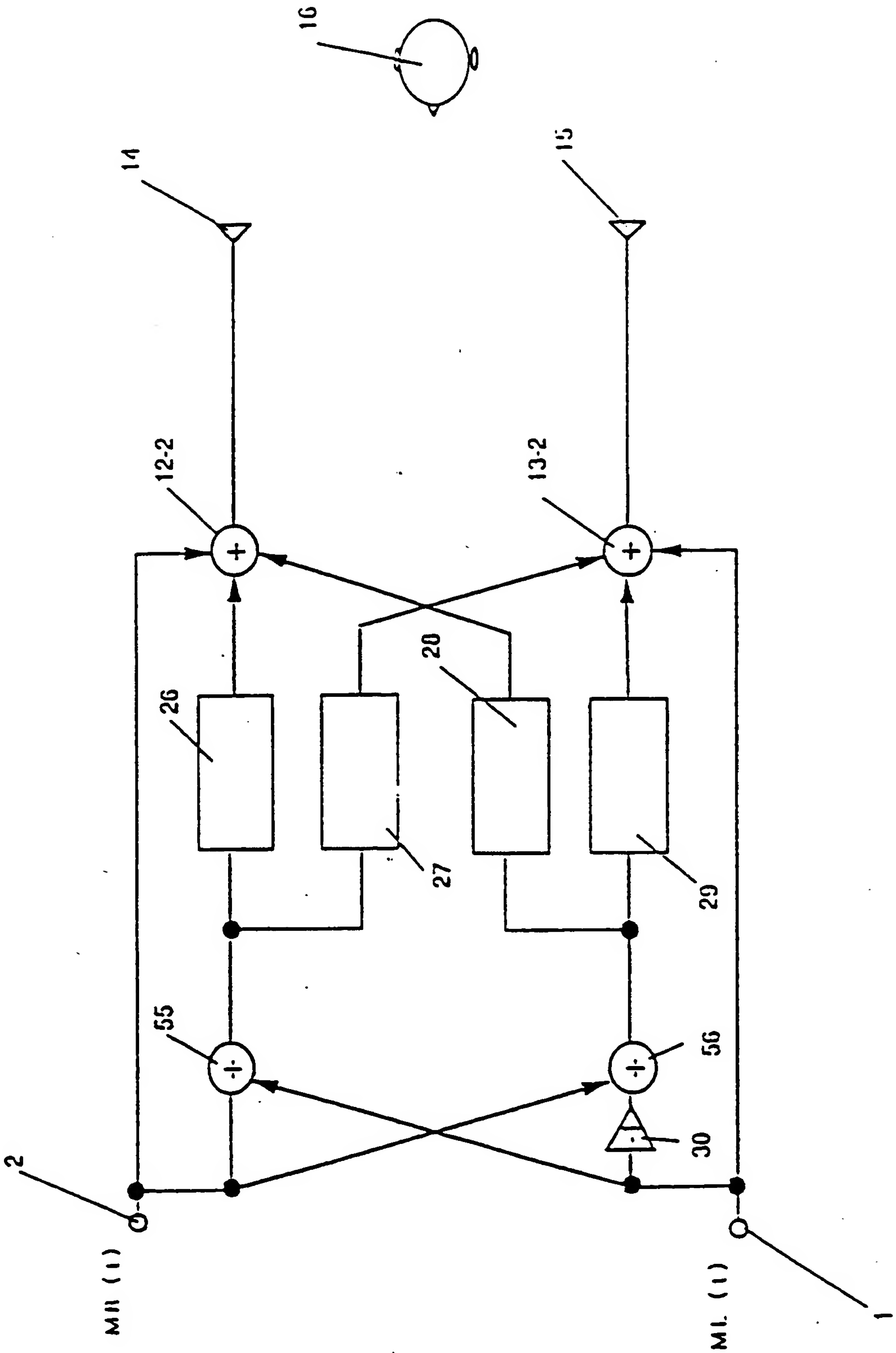
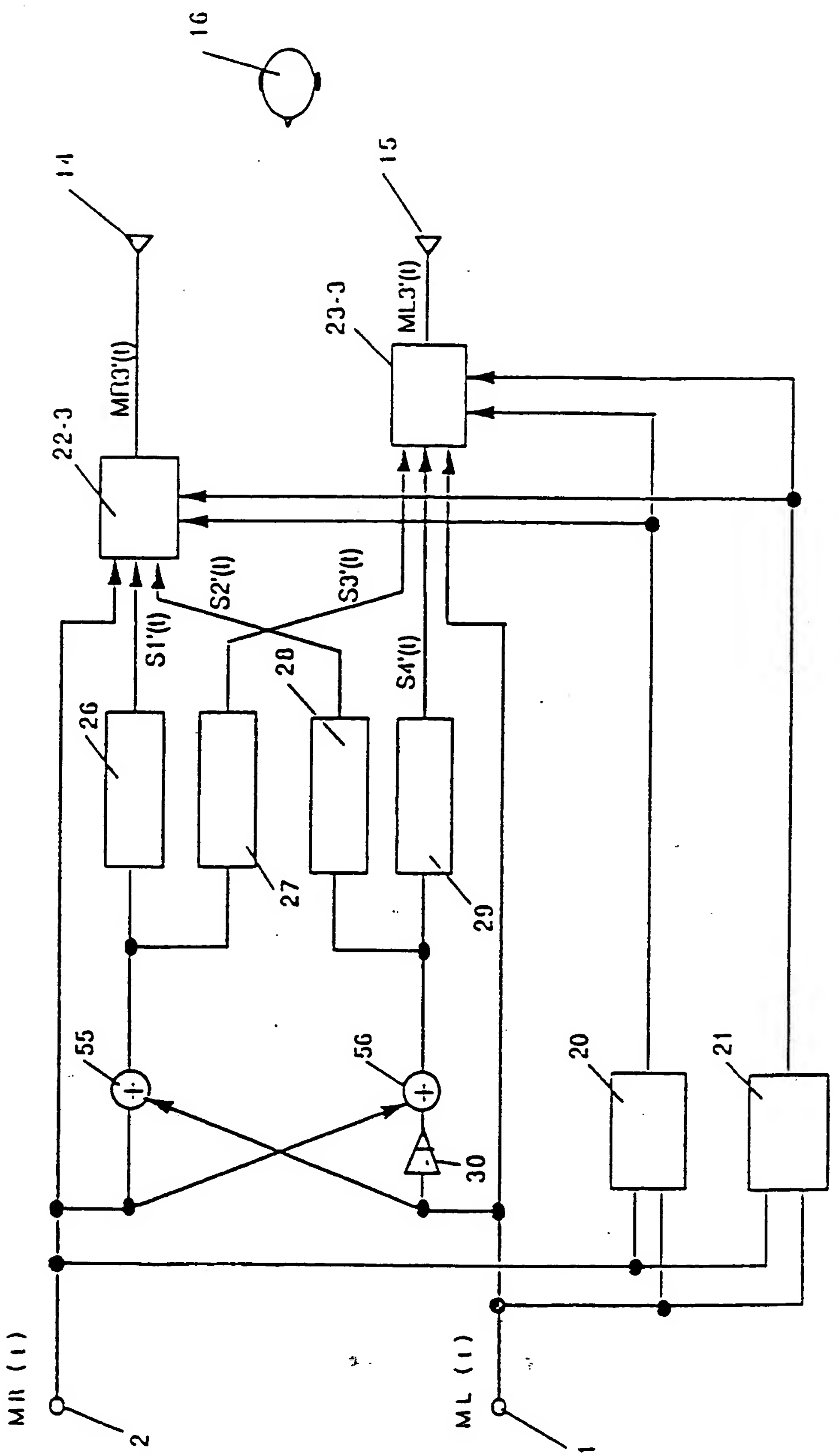


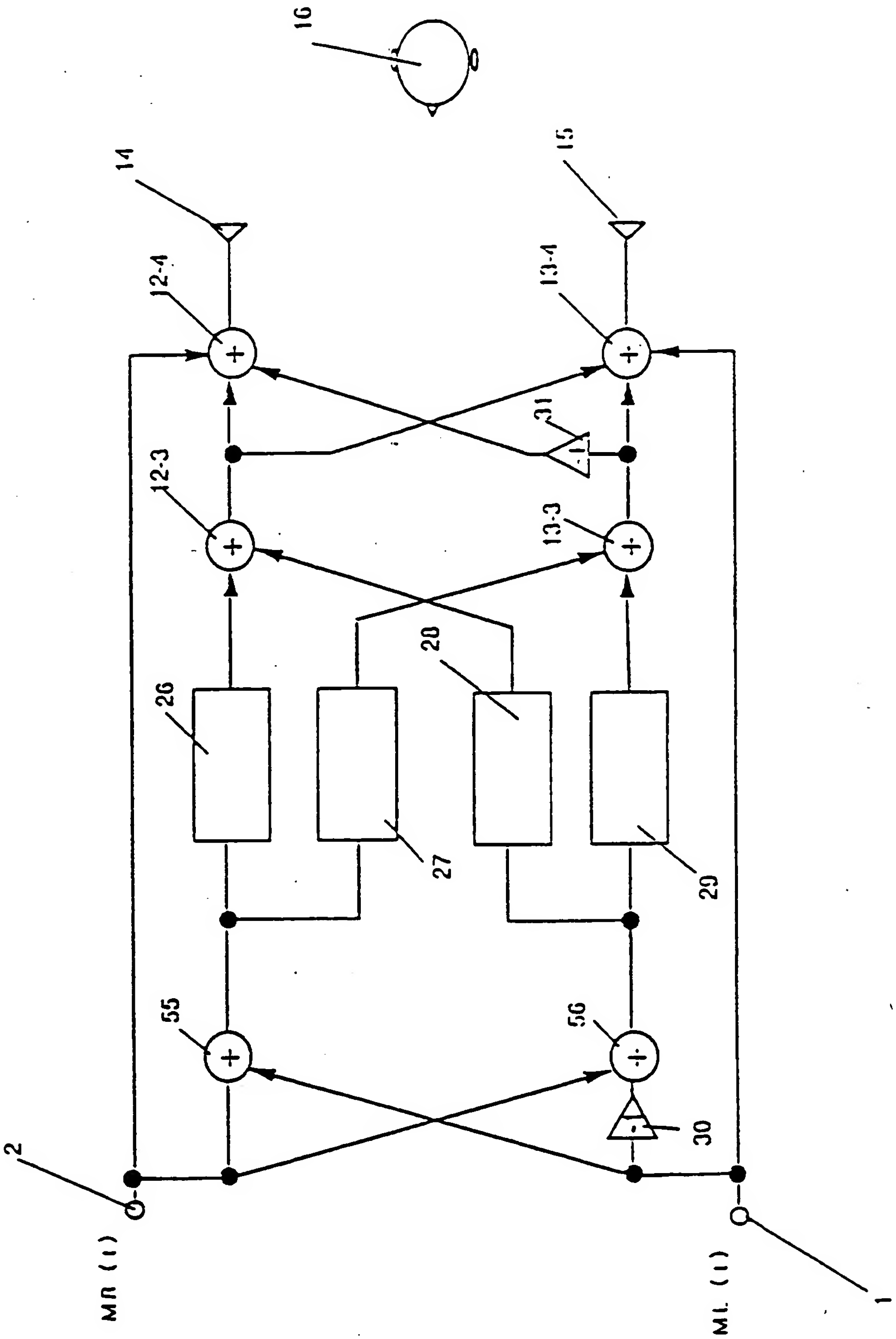
Fig.14



14
15
16
22-3
23-3
S1'(t)
S2'(t)
S3'(t)
S4'(t)
26
27
28
29
55
56
30
20
21
MR (t)
ML (t)
2
1

Fig.15

Fig. 16



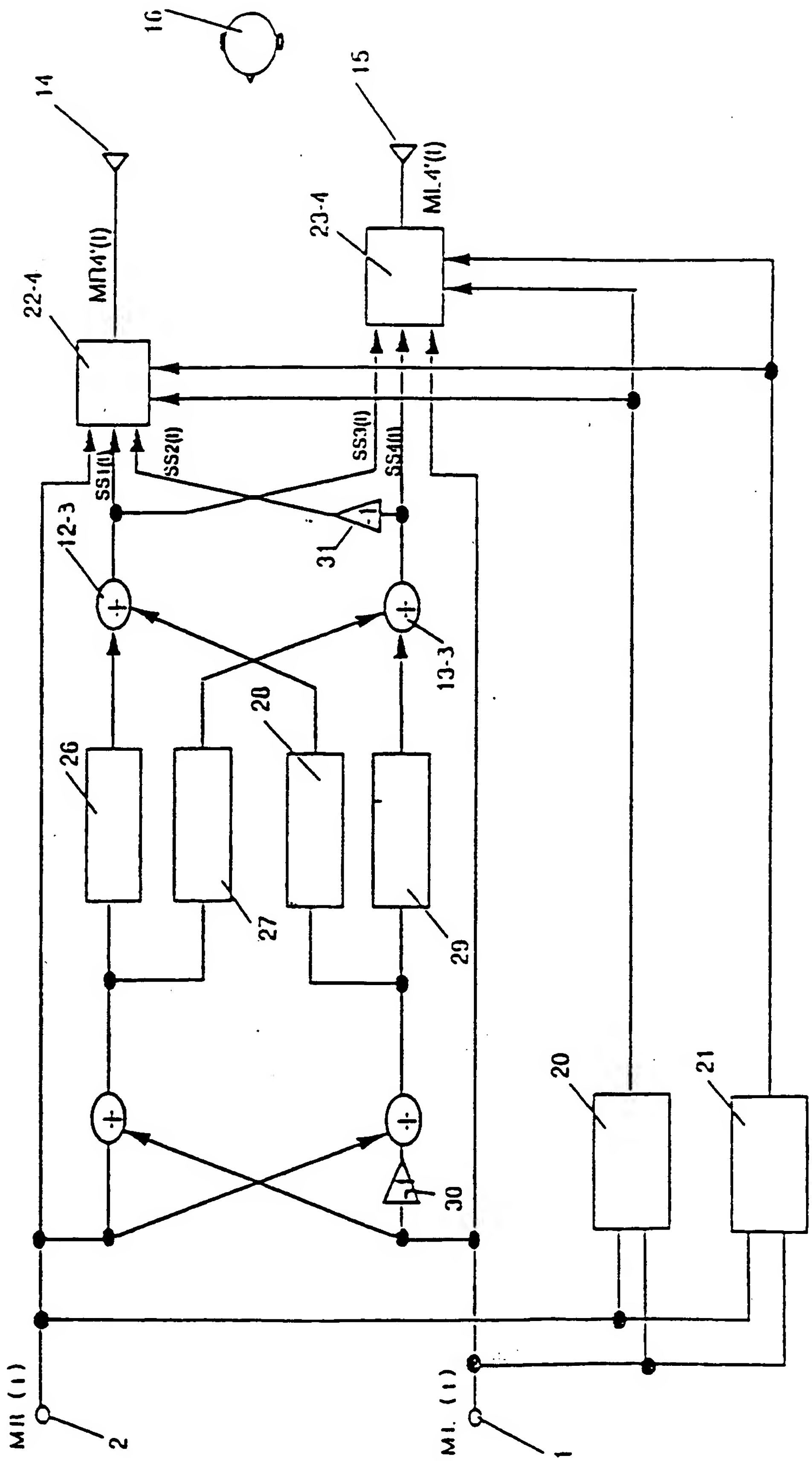

16
14
15
22-4
23-4
MR4'(t)
ML4'(t)
SS1(t)
SS2(t)
SS3(t)
SS4(t)
12-3
13-3
31
-1
26
27
28
29
30
20
21
MR(t)
ML(t)
2
1

Fig. 17

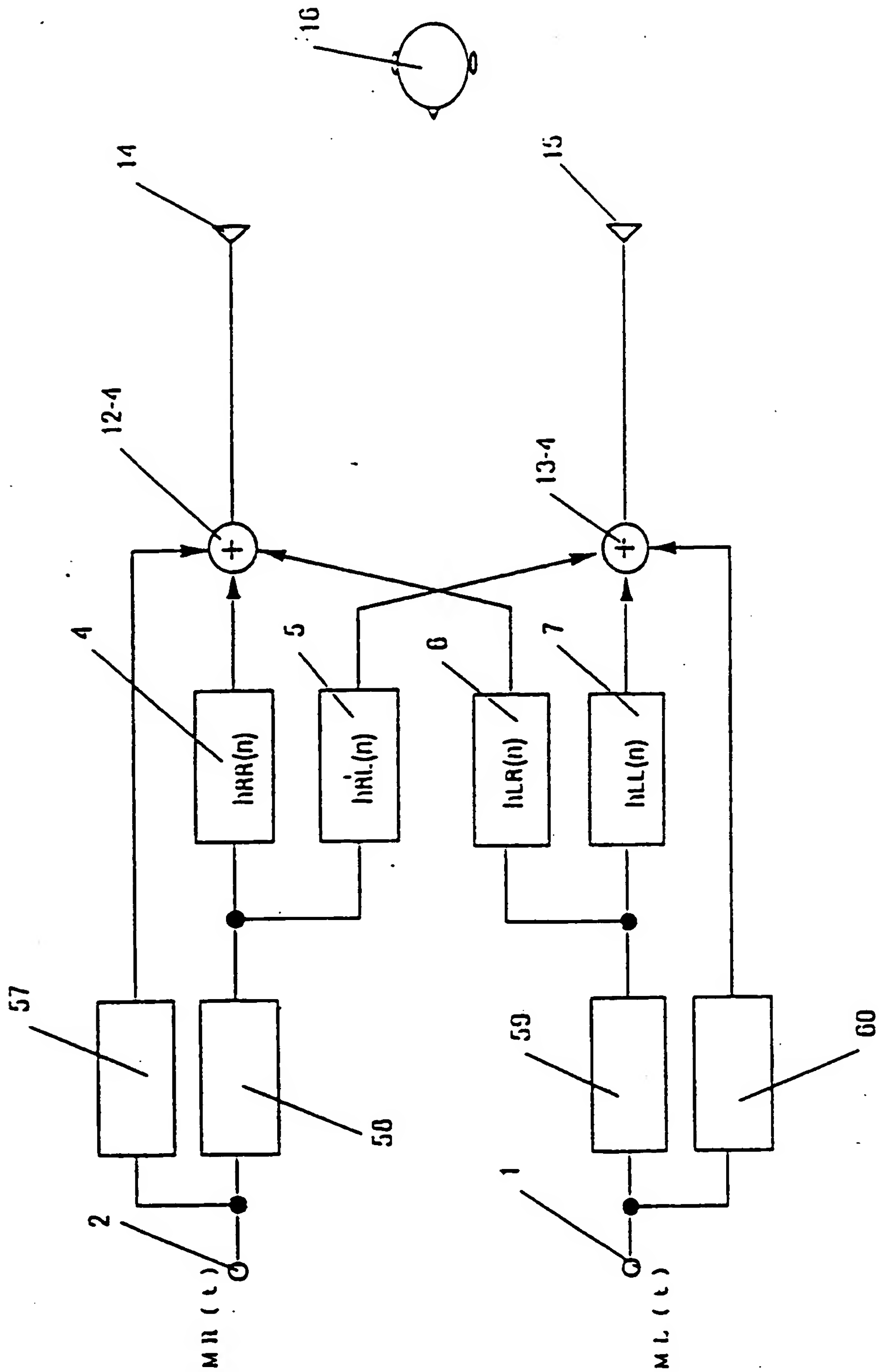


Fig.18

Fig.19A

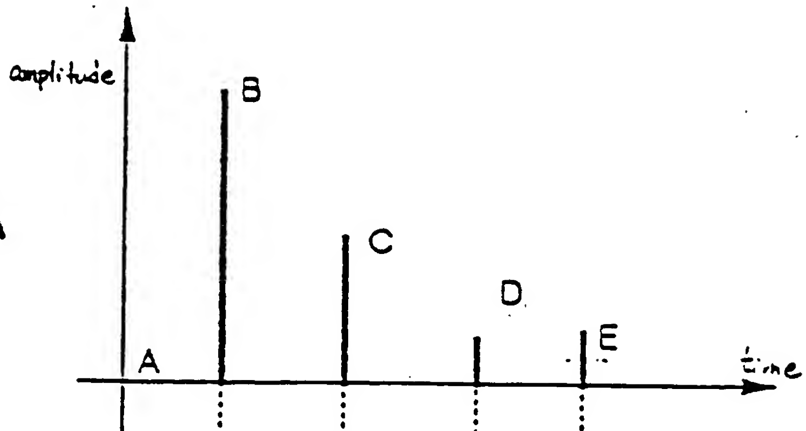


Fig.19B

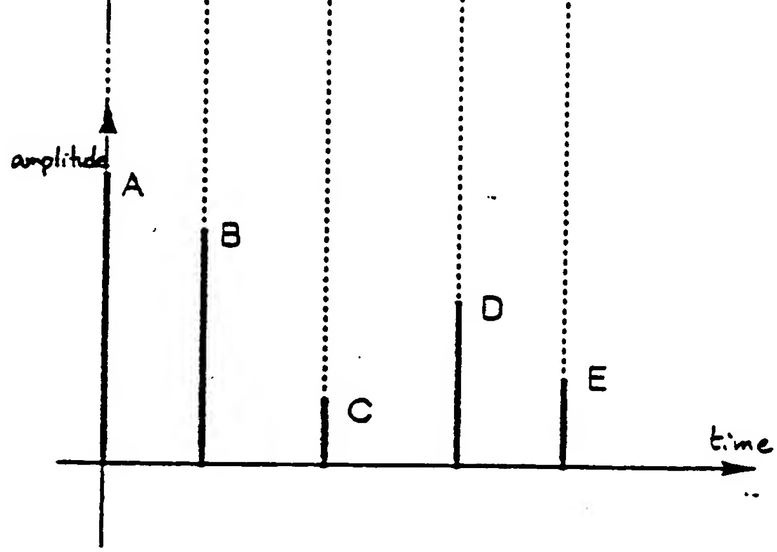
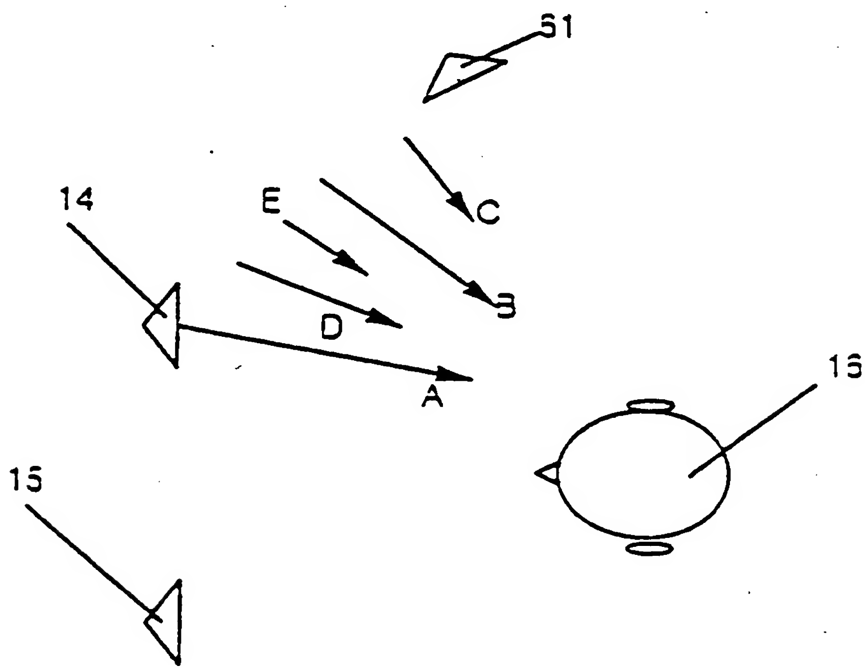


Fig.19C



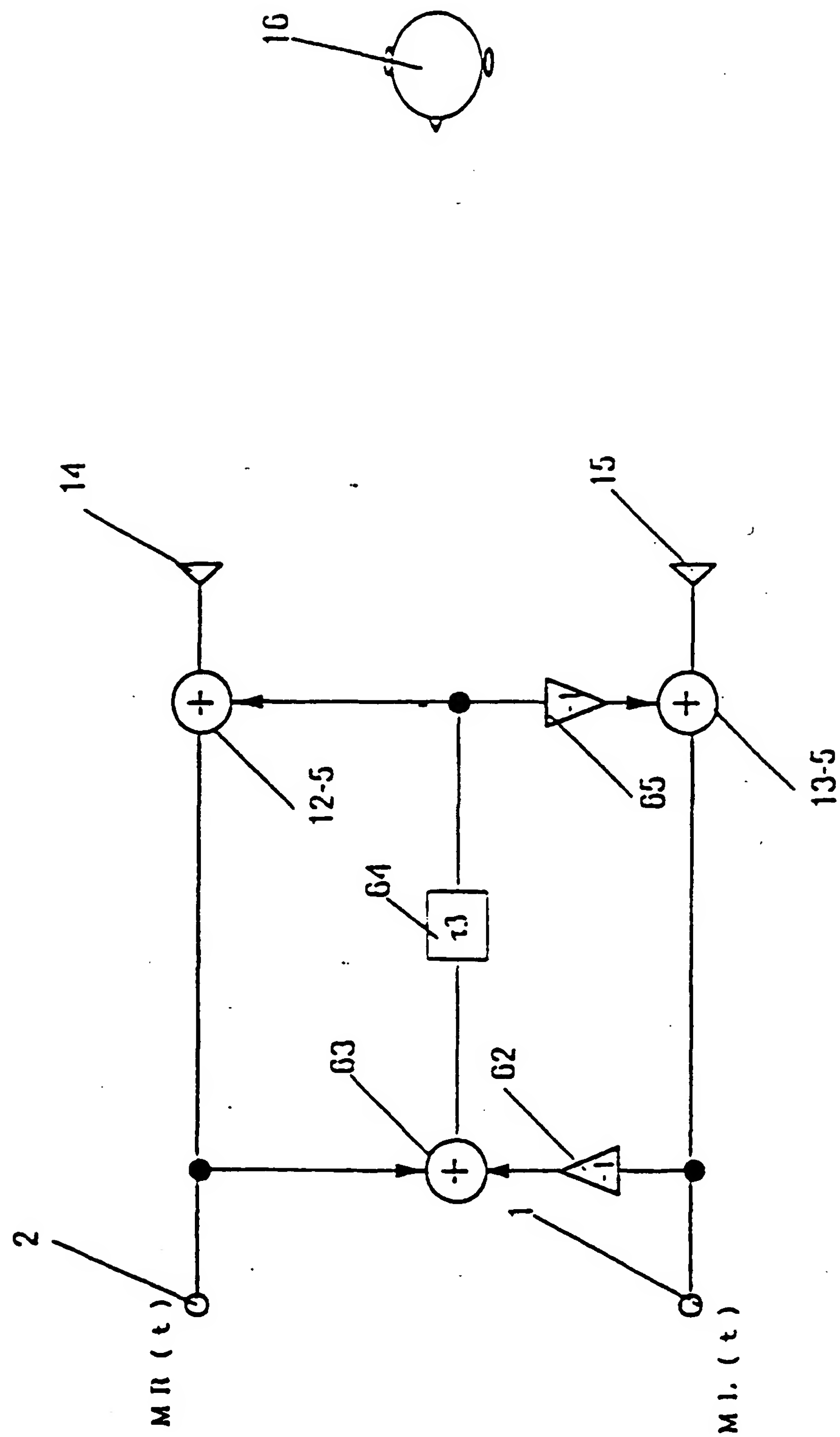


Fig.20

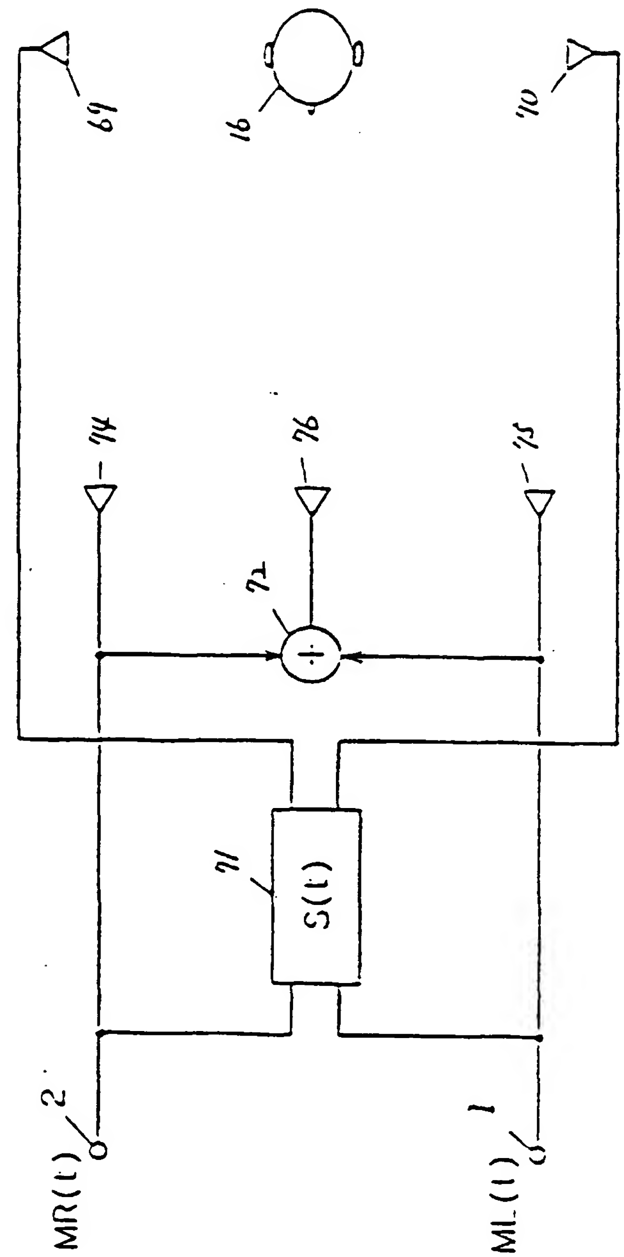


Fig. 21

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 98 10 7944

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
Y	EP 0 422 955 A (MATSUSHITA) 17 April 1991 * column 1, line 1-5 * * page 2, line 29 - page 4, line 38 * ---	1-4	H04S1/00
Y	US 4 706 291 A (KAKUBO ET AL.) 10 November 1987 * column 1, line 5-14 * * column 1, line 35-66 * * column 7, line 29 - column 10, line 56 * * column 11, line 54 - column 14, line 36 * * column 15, line 26 - column 18, line 14 * ---	1-4	
A	PATENT ABSTRACTS OF JAPAN vol. 13, no. 404 (E-817), 7 September 1989 & JP 01 144900 A (MATSUSHITA), 7 June 1989, * abstract * -----	1-4	

TECHNICAL FIELDS SEARCHED (Int.Cl.6)

H04S
H03H
H04H

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	21 July 1998	Zanti, P

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
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